SOLIDYNE
DIGITAL POWER

SERIE 2600
broadcast mixers

USER'S MANUAL

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Getting started — Chapter 1

1.1 What's in the box?

With the console, you will receive the following items:

- This user's manual.
- 1 Interlock power cord.
- Guaranty certificate.
- 1 tubular connector for Tally lights.
- 1 set of self-adhesive labels for channel identification.
- 4 self-adhesive rubber supports.
- 1 screwdriver for adjustment of gain presets.
- spare parts: *2- TL074; 1- TL072; 1- CD4013; 1- NE5532; 1 button MEC; 1 button MULTIMEC; 1 slide potentiometer 10K LIN; 1 rotary potentiometer 50K LIN*

OPTIONALS

- ON-AIR Tally light
- Wiring kit SOL-45
- STUDIO BOX
- STUDIO BOX (headphone distributor/amp)

1.2 Brief description

1.2.1 Input modules

All input modules of consoles *Solidyne 2600* manage the audio signal electronically, that is to say, there is not audio signal on the switches. The faders are ultra-light touch and operate managing the audio indirectly through digital control amplifiers. All input modules are programmable from the front avoiding the use of internal jumpers.

The Solidyne 2600 consoles have a great flexibility since they are fully modular. This feature allows to reconfigure the console without leaving the air. Most the input modules are double. Each **module** has two *faders* of 100mm that manage **two stereo signals**, and each channel has two inputs (Line or Microphone and Aux) commutable from a button in the front panel. There are different modules to manage different signals, that are:

Microphone modules (2610 and 2612)**:** The modules 2610 manage two microphone channels (MIC) and two unbalanced stereo line inputs (AUX). The 2610 features:

- Pan-pot
- Gain control with 30dB of range (+/- 15dB) that allows to adjust the input levels for comfortable operation of the main fader.
- 2 bands low/high EQ, Baxandall type, with +/- 15dB of rage.

The mic module 2612 manages 3 microphone channels and 3 unbalanced line inputs (AUX). It have the same features that the 2610 but include a stage of microphone processor with dynamic compression and noise gate.

Line modules (2601): Manage 2 balanced stereo inputs (LIN) and two unbalanced inputs (AUX). Main and auxiliary gains can be adjusted using 4 trims located at the front panel, to match the levels of the different sources.

Digital inputs 2602: Modules **2602** offers **direct connection to the computer** via **USB**. It works as external sound card, giving to the computer 2 playing devices and 2 recording devices (PGM and AUD can be recorder directly).

2602 modules also allow operation with Skype or telephone hybrids because each internally generates its own Mix-Minus, so the "Solidyne 2600" console has two channels "Mix-Minus" for every 2602

1.2.2 Outputs

All outputs are located on the Master 2607. Line (2601 and 2602) and microphone modules (2610 and 2612) have 3 stereo sends: PGM (program), AUD (audition) and SEND.

- The **PROGRAM** output is the main output, used to send the audio signal to the transmitter.
- **AUDITION** is used to record or to hear audio in control monitors but not on the air.
- **SEND** is a stereo bus. It's used, by example, to make a mix of all input channels, except the microphones. This mix can be sent to the studio monitors. This way the speakers will be able to listen a return in the loudspeakers, without headphones, still on the air.
- The **CUE** bus allows hearing the signals with the channel off and fader closed. The monaural audio hears through an internal speaker, located at the right of the meters bridge. CUE can be assigned to the main monitors of Control Room.

1.2.3 Start devices output

This output modules present in the console (stereo mini-jack located next RJ45) can command the Audicom computer, digital audio processors or any other device with a remote control. Activation is performed by opening the fader and press the AIR button on each channel.

1.2.4 Cue

There are several modes for the cueing:

"Solo": In this mode, when a CUE button is pressed and then another; the second CUE turn off the first one. If all channels are set in mode "soloist"; then only one CUE can be on. By pressing a CUE buton that is on, it will off.

"Mix": CUE works with independence of other channels. The CUE button turns on or off each time that is pressed. All channels defined as "Mix mode" can have the CUE button turned on at the same time.

"Auto-off": With independence of the modes explained above, the option Auto-off turn off the CUE when the channel is send on-air (AIR button turned ON and fader raised). The CUE can be turned on once the channel is on-air, but it will turned off again if the fader is moved.

The CUE behavior is defined in the configuration of each module; as explains at “2.4.2. - Customizable features".

The level of the CUE speaker is set using a knob located at the Master Module.

At the section "Control Room", there is a button called "CUE to Speaker" that allows to send the CUE signal to the Control Room's main monitors and headphones.

1.2.5 ETM-CPU faders

The 2600 uses a new generation of slide potentiometers: conductive ceramic with *“feather touch”*. In normal use, offers more than 20 years of duty. They have a very solid construction, as the skate slides on two sylver-steel cylindrical axis with teflon bushings. The fader generates a DC voltage that controls the module's internal computer. In a 2600 mixer the audio signal never passes through a carbon track neither mechanical switches. For more information please visit our web site.

Main advantages of this technology are:

- Eliminates noise generated by dirty faders.
- Maintenance free and greater useful life.
- Maintains a perfect stereo tracking (error>0,1 dB).
- Allows the use of conductive ceramic sliding potentiometers which offers 2 millions of operations guaranteed.

1.2.6 On-Air mics button

Located above the main fader, this button turn on/off the channel. The button lights when is on.

The button ON AIR switch the signal electronically; there in no mechanical contacts. It gives an operation clicks & plops free, due to 2600's uses a fast slope fading action instead of the hard switch that are characteristic on the mechanical switches.

When an AIR button is pressed, turns on the **tally light** (the console gives 12 V DC as tally signal). Also, a relay **mutes the studio monitors**.

A Master Microphone button is located on the 2607 Master module. The purpose of this button is switch on/off all microphones channels at the same time. Each channel can be linked to Master Mic button with independence. By default, all mic channels are linked to the Master Mic button.

1.3 Expansions & accessories

The 2600 series can be customized according to the requirements of the radio station. The user decides how many microphone and line modules will have.

All models feature as an option the kit of wiring and adapters for connecting the console without the need to assemble cables.

You can also choose to include other optional features, which are described below:

	2600 XL	2600 XX	2600 XD	2600 XZ
INPUT CHANNELS (MIC 2601, 2612; LIN 2610, 2602)	Up to 16	Up to 26	Up to 16	Up to 28
AES-3 OUTPUT	OPTIONAL	OPTIONAL	OPTIONAL	YES
STREAMING OUTPUT (AoIP)	OPTIONAL	OPTIONAL	YES	YES
VC180	YES	YES	YES	YES
Clock-Counter	YES	YES	YES (LCD)	YES
Remote Control LAN (/VI)	OPTIONAL	OPTIONAL	OPTIONAL	OPTIONAL

1.3.1 VU and stereo phase vector

The Stereo Phase Vector allows measuring the stereo phase relationship and therefore the grade of stereo sensation achieved. The range is 0 degrees (mono) to 90º (maximum stereo); with inverted phase indication (180 grades), to prevent to the operator that a connection’s error exists in the microphones, or among equipment’s with analogical connections

The analogical section is gauged in steps of 20 degrees, with quadrant of vectors. This allows following the quick program variations. By other hand, the digital section retains for an instant the maximum value reached by the phase rotation.

A musical program with good stereo sensation will have averages (analogic) of 40 to 60 degrees, with peaks (digital) of 80 degrees. Readings of 0 and 20 almost indicate a monaural signal. The indication -0 or INVERTED implies the serious problem of inverted phase. This must be quickly corrected because the inverted signals 'disappear' when they are listen in mono FM receivers.

1.3.2 Timer / Clock

a) Provides the **current time** while the microphone channels are turned-off.

b) Shows a **count up** (minutes: seconds) when the microphones are on the air.

The timer mode is **activated** when the mics are activated using the Master On-Air button**.** When you turn off the microphones, the time will be retained on screen by 3 sec and then the display will return to 'clock mode'. If you turn on the microphones again (in this 3 sec window) the timer will resume.

1.3.3 2630-VQR module

This optional module allows partially restore the voice quality that has been degraded by telephone transmission. Can be used with the internal telephone hybrid , or with external hybrids connected to the console via "External Hybrid" connector.

The 2630-VQR module has three 100 mm faders that allow to the operator adjust the degree of reconstruction applied to the audio; and a LED indicator showing restoration levels. The module also features an noise gate of variable threshold.

Refer to "3.6 Module 2630 VQR" for the operation of this module.

1.3.4 AoIP output

The output "AoIP" streams the program mix to another console Solidyne 2600. This streaming also be decoded with any streaming player software; or by using a Solidyne ADA102 decoder.

1.3.5 AES-3 outputs (AES/EBU)

Optionally, models XL & XX can have AES-3 outputs for **program** and **audition**.

1.3.6 STUDIO BOX

Studio Box is an optional accessory which concentrates in a unique box all monitoring facilities into the Studio. Studio Box offers the following features:

- **5 headphones** outputs with independent control level.
- Output for **Studio Monitors** with level control.
- **Tally**
- **Timer / Clock**
- **Talkback**, to talk with the Control Room operator.

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Setup and connections — Chapter 2

2.1 About the wiring

The consoles come with their inputs and outputs with RJ45 connectors and are wired using multipair cable shielded CAT-5. With the advent of audio over IP (AoIP) various manufacturers started using RJ45 connectors and shielded multipair cable to replace the various audio connectors, to standardize the entire installation with a single type of connector and a unique type of cable.

Additionally, the use of structured cable for connection between remote devices facilitates the installation anywhere in the world, due to availability of components and assembly tools used in data networks. The multipair cable end that connects to the audio device (microphones, speakers, playback devices) still require standard audio connectors. For this, the RJ45 cable provides termination sections with RJ-45 female at one end and connector audio as needed at the other.

2.1.1 Solidyne SOL45 wiring kit

Solidyne provides (as option) different kits of cables "**SOL45",** designed to make the installation of a Solidyne 2600 console in a few minutes. This wiring kits are made using shielded multipair CAT-5.

2.1.1.1 Standard kit

The basic option is "SOL45 / 10" that connects consoles up to 10 audio channels. Consolas with more than 10 channels will use the "SOL45/NN" kit being "NN" the number of audio channels.

Consoles wired with these kits will have the connections detailed below. But if you need to add more cables, you can do so freely as explained below.

NOTE: in all cables the channel marked with a red label is the right channel.

1 MIC & LIN analog inputs. By each MIC or LINE channel there are:

- One RJ45-Male to RJ45-Male; length 2,5 m + one Adapter RJ45-Female to 2 x XLR-Female (Left & Right) In **MIC cables ONLY the Left is used.**
- In line modules (2601) will be provided a half of cables with XLR-F and the other half with RCA connector.

2 USB inputs: For each module 2602 are included:

- 2 USB cables; length 1,80 meters

  NOTE: If using Solidyne's DH400 Telephonic Hybrids, or another hybrid with independent mix-minus, ask cables SOL45.

3 Outputs of Master 2607

2 cables RJ45-M to RJ45-M, length 2,5 meters + Adapter RJ45-F to 2 x XLR-M (channels Left and Right) (One cable to balanced PGM output and another cable for balanced AUD).

1 cable RJ45-M to RJ45-M length 2,5 meters + Adapter RJ45-F to miniplug stereo (SND unbal output).

1 cable RJ45-M to RJ45-M, length 2,5 meters + Adapter RJ45-F to 2 RCA (input for external tuner).

Note 1: if the console have the optionals /VI, /AoIP each optional includes its cable (2,5m.

Note 2: if the console have the optional /AES comes with a cable of 2,5meters + adapter to 2xXLR (one for AES- PGM and another for AES-AUD).

2.1.1.2 Additional cables

If additional cables are needed, the options are:

1. SOL-25 /SOL-50 Cable RJ45-M to RJ45-M, flexible shielded, length 2,5/5 meters.

   Note: Longer cables can be used using CAT-5 shielded multipair and shielded RJ45 connectors.

2. SOL-XLRM Adapter RJ45-Female to 2 XLR-Male connectors (balanced)

1. SOL-XLRF Adapter RJ45-Female to 2 XLR-Female connectors (balanced)

2. SOL-MINIM Adapter RJ45-Female to 1 Miniplug 1/8" stereo (unbalanced)

3. SOL-TRSM Adapter RJ45-Female to 2Plug ¼" TRS balanced (separated Left and Right).

4. SOL-RCAM Adapter RJ45-F to 2 RCA connectors (unbalanced)

5. SOL-TAIL Adapter RJ-45-F to 8 cables (Bare Bone) ready to weld any type of connector..

2.1.1 Parasitic signals

IF THE WIRING OF THE CONSOLE HAVE INTERFERENCE FROM EXTERNAL RF SOURCES, PLEASE READ THIS SECTION.

Generally, are considerate parasitic signals all unwanted signals that appear in audio lines. A common type are denominated *humming*, low frequency signals (multiples of 50/60 Hertz) caused by the interaction of electromagnetic fields coming from the AC line. When the interference source is a magnetic field (generally originated in a supply transformer) the resultant interference will be denominated *electromagnetic humming*. When the interference is due to such electric potentials as cables that take supply tensions, that are elevated in comparison with the audio signals present in the circuit, you will be in presence of *electrostatic humming*. The distinction is not merely academic, because the resolution of a problem supposes the knowledge of the noise type to apply the correct solution.

Examples: To minimize the reception of electromagnetic humming in the wires, remember the following rule: "THE AREA AMONG TWO AUDIO WIRES WILL BE MINIMUM." It implies that the cables will be tied very close, like the shielded twisted pair audio cables. They should pass far away from any transformer or devices that manage high-intensity currents. Is important to remember that a wire can be good shielded, but if his conductors don't complete the conditions of minimum area it will be susceptible to take magnetic humming.

Other parasitic signals are: AC HUM, RADIO FREQUENCY and CROSSTALK. As hum noises like the radio frequency are originated by electromagnetic fields of high frequency; the first ones are originated by disturbances due to the connection and disconnection of equipment's to the AC line, the seconds ones are generated by communications transmitters or industrial equipment. If these signals penetrate into audio lines, with sufficient intensity, can surpass the action of the special protection filters, and to reach some sensible part of the input stages. In that case, the interfering signals can be demodulated and already turned audio signal, and will be amplified by the rest of the system. It is fundamental, therefore, to maintain the interference within reduced margins. It is obtained avoiding very long audio lines, with aerial sections or that pass near of transformers or RF transmitters. For protection against very high frequencies is advisable to use double shielded cables, guarantied by the manufacturer

CROSSTALK is the reception of signals coming from other lines of audio. This, like all unwanted noise, it can be supposed controlled when its level is below the level of the system residual noise. Then, all considerations mentioned for the case of buzz are valid.

2.1.2 RF interference

2600 consoles have numerous internal protections against RF fields, for the AM and FM broadcasting band. When the transmission station is installed correctly, there will be no interference problems, still with FM equipment of 50 kW installed in the terrace of the radio station. Nevertheless, when the antenna is badly positioned with respect to the Studio or has severe SWR problems, then it does not have the minimum value of field intensity, downward. Or maybe there is a faulty ground connection, then, strong standing waves will appear on the cables of the Studio that can induce high electrical currents inside the audio console.

Symptoms: If the interference takes place at the A.M. band, the sound transmitted by the A.M. radio will be listened in the loudspeakers on background (or at buses PGM, AUD or SEND). In case of FM transmissions, the interference inside the console demodulates the A.M component of the FM carrier; (usually hum from power supply) causing background humming, because in many transmitters of FM, the final output stage is not powered with stabilized tension. Therefore, if console 2600 presents humming, please make a test shutting down the transmitter a few seconds to check if the problem disappears. Sometimes, an FM transmitter with the output stage badly calibrated also produce interference in which you can hear the transmitted audio (due to the misalignment a part of the FM modulation is translated to AM modulation).

Solution: 2600 consoles already have internal filters to avoid the RF interference. Therefore, if interference's appears, normally is caused by installation problems on the transmission station that generate elevated currents in the wiring of audio on the Studies, due to the standing waves. These currents circulates inside the cabinet of the 2600, and enters to the console when overpass the barriers that impose to this currents the built-in RF filters. The solution, then, must be external to the console.

The solution is to acquire **ferrite O-rings** of 60 mm diameter, to allow passing the cables and connectors. They will be used in each one of the D25 connectors, in the inputs and outputs. The total set of cables that goes to each connector, will have to be twisted with a complete return around the ring (see 2.3.3 - Wiring).

2.2 Inputs & Outputs connections

2.2.1 Rear Panel

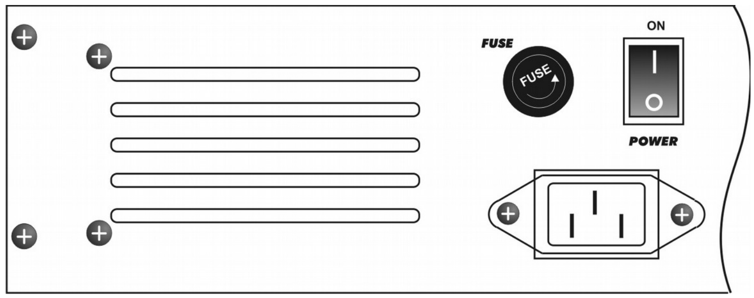


Fig.3a Rear panel– power source

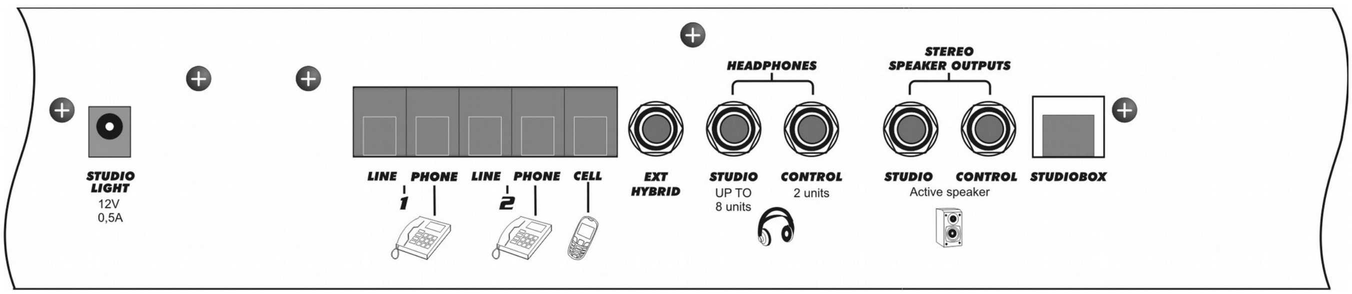


Fig.3b Rear panel– connectors

At left, the console's rear panel have the on/off switch, the main fuse (1 Amp) and the AC connector (Interlock power-cord, factory provided). The console can be used with 110/240 VAC 50/60 Hz, with automatic selection (switching power source).

The rear panel also contains the connectors for the loudspeakers monitors, headphones, and telephone lines with the associate phone sets and the output to plug the ONAIR light (Tally).

All inputs connectors are located under the rear panel, using RJ-45 connectors. Remember: good connections between the console and audio equipment are very important question to guaranty a sure and free of flaws operation. For that, we recommend you take the necessary time to carry out the wiring of the console, with a great care and always using high quality components. A good solution is to acquire the wiring kit **Solidyne SOL45.**

2.2.2 MONITOR OUTPUTS

The outputs "STUDIO" and "CONTROL ROOM" are of line level, designed to be used with powered speakers. The connectors are TRS 1/4”. The output level adjusts from Master module with the knobs “Sudio” and “Control room” respectively.

2.2.3 HEADPHONES

STUDIO and CONTROL ROOM have independent outputs, using ¼” jacks connectors. The STUDIO output allows to connect up to 8 headphones. CONTROL ROOM output supports up to 2 headphones. The headphone outputs have protection against accidental short circuits.

At the console, the operator has independent level controls for Studio and Control headphones. We recommend to install at the Studios the Solidyne Studio Box (optional): a headphones amplifier/mixer. This unit is placed on the table and manage up to 5 headphones with independent volume controls. In addition StudioBox offers talk-back to the control room.

2.2.4 STUDIO LIGHT (TALLY)

The STUDIO LIGHT output gives **12 volts / 0,3 A** when one or more microphones are on-air. This allows to connect up to 5 LED’s lights (12V/60mA).

As alternative, the outputs “Starts Devices” can be used to manage individuals lights for each microphone. The electric circuit is shown in the general connections diagram (please see the 2.2.2.4).

FUSE: The on-air light is protected by an internal fuse of 1A. If the fuse blows, you must to remove the rear panel to replace it. This fuse is located near to the tubular tally connector.

2.2.5 CONNECTING THE PHONE LINES

The **Hybrid 2607** manages 2 telephone lines and 1 cell phone input. Once connected, the rejection factor must be re-adjusted to be adapted to the local telephone line impedance (please see 3.2.1 – Hybrid 2607)

At the rear panel there are five RJ-11 connectors. Two are for telephone lines inputs; two are for connecting the associated telephone sets (Production Phones) and the fifth is for the cell phone. The lines can be connected directly to the radio station's PBX or directly to the public telephone system.

The hybrid inputs have internal filters for RF rejection, effective in AM, FM and UHF bands. The telephone lines have internal high tension protection using Metal Oxide Varistors. But we recommend having additionally a good external protection.

In case of damage on the internal fuses, these can be ordered. See Chapter 4

Phone lines protection

The telephone lines must have protection against lightning and electrical surges. Please, see general protection diagram at the end of this chapter.

2.2.5.1 Connecting a cell phone

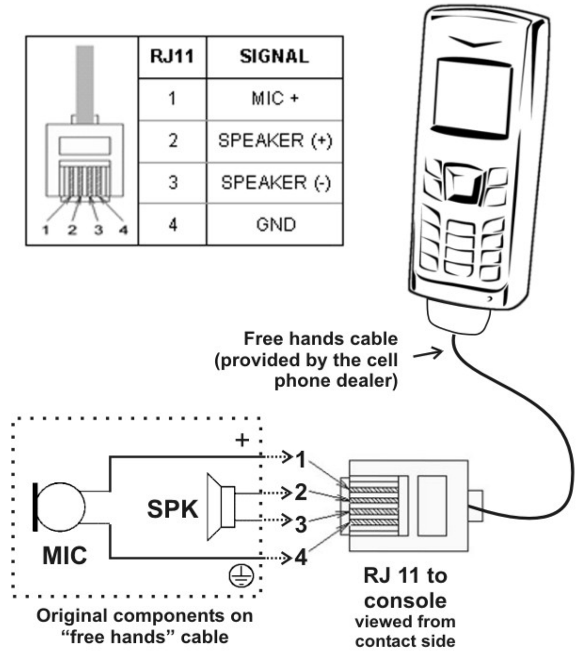
The cell phone connects directly to "CELL" RJ-11. An adaptation cable is required for connect the cell phone to the console. This adaptor defers according to mark and model of the cellular. The cellular connects to the hybrid using the "hands free" connection. Consult with a Solidyne dealer by this accessory. For make this yourself, please read the following indications.

Will need a **special cable RJ-11** whose connection will depend on the brand and model of the cell phone. Purchase the "hands free" accessory that correspond to the cellular and consult the documentation to make the connection (see next image).

Basically, which transmits the cell phone through the "hands free" mode is the audio signal of the cell telephone: microphone and loudspeaker. The Hybrid receives, via cell phone, the remote audio (that is to say, the audio of who is talking at the other end of the line). On the other hand, the hybrid sends to the cell phone the audio that originates at the studios (return signal).

Usually, microphone and loudspeaker of the cell phone are disconnected while "hands free" mode is used.

RJ11	SIGNAL
1	MIC +
2	SPEAKER (+)
3	SPEAKER (-)
4	GND



2.2.5.2 Connecting an external Hybrid

The female TRS ¼" labeled "EXT HYBRID" gives PGM send (tip) and return from Hybrid (ring), to connect an external Hybrid. The PGM send is Mix-minus: it includes all the signals of the PGM Bus, with exception of the *Return from Hybrid*, to avoid a feedback loop.

The signal that is incoming by the external hybrid's return, is routed to the same circuit of control that the signals of the internal hybrid. Therefore, the "Hybrid Fader" will behave in the same way that it works with the telephone lines connected to the console; sending the signal to the air or to CUE according to the fader position. When connects an external hybrid, the internal hybrid remains active and can be used with independence of the external hybrid.

The lines connected to console and the external hybrid can be in conference.

Remember: once a call is on-air, the cue channel stays disconnected, so the user must use a phone associated with that line to speak privately with the second caller.

When make a conference between internal and external hybrid, **take care with the output level of the external hybrid**, so an excessive level can cause feedbacks.

2.2.6 "STUDIO BOX"

The RJ45 on the right (Master Module back side) provides connection to the accessory Solidyne Studio Box. Studio Box is the solution for sending all monitoring signals to the study. Includes distribution amplifier to connect up to five headphones, connection for speakers, and Talk-back system for commu-

nication with the control (see 2.5.3 - Studio Box). It uses a standard UTP cable.

2.2.7 INPUT MODULES

2.2.7.1 Line inputs

The line inputs are stereo, electronically balanced (transformerless). It allows the connection of high-level signals, balanced or unbalanced. Input gain adjusts from the frontal panel with the GAIN trims (L & R). In order to unbalance this inputs, connect the negative (-) pin to GND and positive pin to signal (see *"2.3 – Connectors and wiring"*).

AUX inputs are unbalanced stereo. It manages -10dBu / 10KOhms and +4/+8 dBu line levels. Input gain adjusts from the frontal panel with the GAIN AUX trim (L&R).

The impedance of LINE and AUX input is bigger than 10 Kohms (Bridging inputs). This is compatible with all current equipment (sound cards, CD-player, audio interfaces, etc.). But if you need to adapt the input to 600 ohms, place inside the connector a 680 ohms resistor in parallel with the input.

2.2.7.2 Microphone inputs

Microphone inputs are electronically balanced. The gain adjusts from the frontal panel with a trim control with 30dB of range. The pan-pot allows change the microphone position in the stereo image. AUX inputs are similar to line modules, but don't have gain presets. The MIC modules use RJ45 connectors for mic and aux inputs. Please see connections at *2.3 'Connections and wiring'*.

2.2.8 The Master Module

PGM and AUD outputs are electronically balanced, with a high **reject** to common mode signals. To unbalance a balanced output, only connect the positive pin (+), and leave unconnected the negative one (-). SEND output is stereo, unbalanced.

The **nominal output level** is: 0VU = +4dBm. When a balanced output is connected to unbalanced input, its level is reduced in 6 dB, therefore 0VU= -2dBm. Notice that if some nonprofessional equipment connects to the 2600's output, possibly the level will be too high and some clipping will occur. In this cases use a resistor attenuator to reduce the level.

ON-AIR input: This is a stereo input to connect an **external tuner**. Both the studio and control room, must be heard the tuned air transmission, because the transmitted audio is processed and therefore sound levels change with respect to the signal on the console.

Use professional tuners with **outputs of low impedance** (600 Ohms). In case of use consumer tuners, you must add a line amplifier between the tuner outputs and the "Air" input of Master 2607. This line amplifier is for adapt the tuner output to low impedance.

CUE input: Master 2607 have a CUE input, designed to connect an output of a computer. This way is not necessary to use additional speakers next to the computer.

This input requires a special cable that connects to the Master's RJ45 "On air in/Cue In" (see "2.3.3 – Additional signals of Master 2607").

2.3 Connectors and wiring

2.3.1 General recommendations

a) Consider to buy the wiring kits SOL45 (2.1.1 "Solidyne SOL45 wiring kit").

b) To arm the cables, use shielded RJ45 and cable of good quality. Remember to join the connector's shield with the shield of the cable.

c) Avoid the cables are hanging from the connector. Use cable guides to distribute cables.

d) Avoid to mix cables with audio with AC cables. Use different ways for each case.

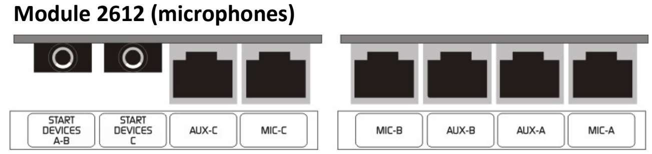
e) If you hear background hum, it can be produced by RF stationary waves, induced from the FM antenna. In this case, you must install ferrite o-rings (60mm diameter) in all input & output cables like shows the left image (see 2.1.2 – RF interference).

2.3.1 Inputs and outputs

Module 2601 (lines)

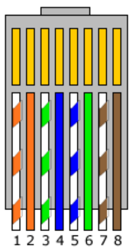
Module 2610 (microphones)

Module 2612 (microphones)



Next tables are valid for all inputs and outputs, balanced and unbalanced.

RJ-45



RJ45 NOMENCLATURE

PIN	WIRE COLOR
1	Orange/White
2	Orange
3	Green/White
4	Blue
5	Blue/White
6	Green
7	Brown/White
8	Brown

BALANCED INPUTS / OUTPUTS

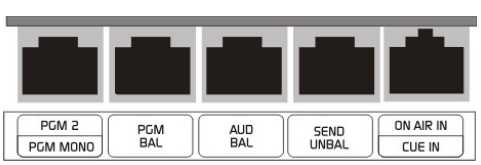
PIN	WIRE COLOR
1 Left channel (+)	Orange/White
2 Left channel (-)	Orange
3 Right channel (+)	Green/White
4 GND	Blue
5 Reserved	Blue/White
6 Right channel (-)	Green
7 -15 (optional)	Brown/White
8 +15 (optional)	Brown

UNBALANCED INPUTS / OUTPUTS

PIN	WIRE COLOR
1 Left channel (+)	Orange/White
2 GND	Orange
3 Right channel (+)	Green/White
4 GND	Blue
5 Reserved	Blue/White
6 GND	Green
7 -15 (optional)	Brown/White
8 +15 (optional)	Brown

2.3.3 Additional signals on Master 2607

The following features requires the use of special cables, which are different from the cables used to the balanced and unbalanced inputs/outputs.



BALANCED OUTPUT PGM-2 + MONO

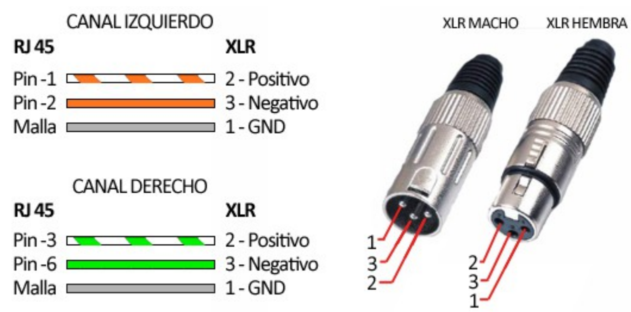
PIN	WIRE COLOR
1 Left channel (+)	Orange/White
2 Left channel (-)	Orange
3 Right channel (+)	Green/White
4 GND	Blue
5 PGM-MONO	**Blue/White**
6 Right channel (-)	Green
7 -15 (optional)	Brown/White
8 +15 (optional)	Brown

ON-AIR MONITOR + CUE

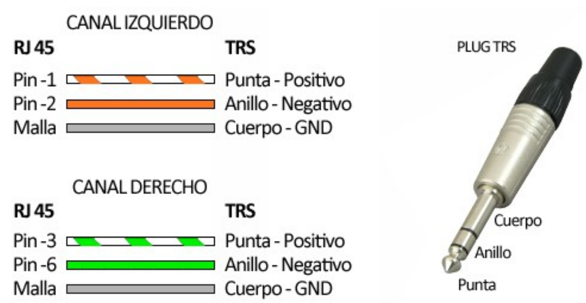
PIN	WIRE COLOR
1 Input ON-AIR **L**	Orange/White
2 No connected	Orange
3 Input ON-AIR **R**	Green/White
4 GND	Blue
5 Input CUE	**Blue/White**
6 No connected	Green
7 -15 (optional)	Brown/White
8 +15 (optional)	Brown

2.3.4 RJ45 to standard connectors

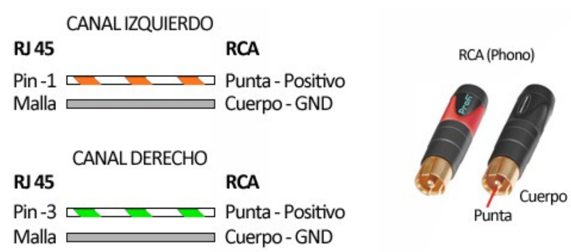
2.3.4.1 To a balanced XLR



2.3.4.2 To a balanced TRS

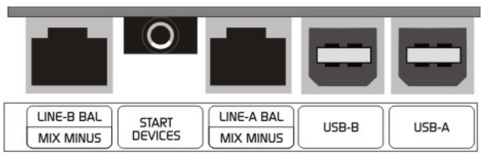


2.3.4.3 To an unbalanced phone-RCA



2.3.5 Module 2602 (USB)

The 2602 module connects directly to the computer via USB. Each channel is recognized by Windows© as USB external sound card. In this way, the audio is send directly from the computer to the console, without leave the digital domain. In addition, the 2602 provides **two analogical balanced inputs**. Balanced inputs are available on RJ45 connectors.



BALANCED INPUTS + MIX-MINUS

PIN	WIRE COLOR
1 Left channel (+)	Orange/White
2 GND	Orange
3 Right channel (+)	Green/White
4 GND	Blue
5 Output mix-minus	**Blue/White**
6 GND	Green
7 -15 (optional)	Brown/White
8 +15 (optional)	Brown

2.3.5.1 USB connection

Each USB channel is independent and can connect to a same computer or two different computers. Both channels are compatible with USB 1.1 and 2.0.

Connect the module "2602" to any USB port of a computer running Windows©. When connecting it, Windows© recognizes the "2602" and install the drivers. Additional drivers are not required. Each channel will appear on Windows like one stereo USB audio device and one stereo USB recording device.

When a USB channel is plugged, the display on the module will show **"PC4"** indicating that one stereo play channel and one stereo recording channel are active. If the second USB channel is connected, the display will change to **"PC8"**; indicating that the eight channels were recognized (2 stereo outputs and 2 stereo inputs).

To check the available sound devices, go to *"Windows Control Panel > Sound devices > Audio".* Here the default sound devices are defined. Remember to update the settings on the audio applications.

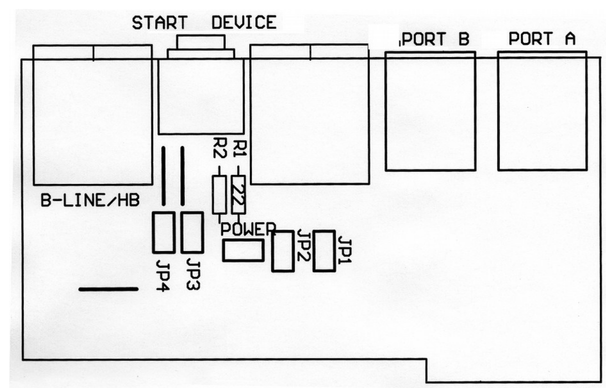
In addition, the module 2602 **offers two sends by USB** that appears on Windows as "USB recording devices". 2602's Channel-A gives the program signal (PGM) and the 2602's Channel-B sends the Audition signal (AUD). The USB-PGM out can be used, for example, for web-casting.

The **level of the digital signal** is set according to the standard **K-15**. The 0 VU (+4 dBu) corresponds to **-15 dBfs** at USB output. There are 15 dB of headroom before clipping (above +4 dBu).

2.3.5.2 *Mix-minus* to use with Vo-IP software

2602 modules have the ability to cancel its own signal on sending PGM USB (mix-minus). This allows to use the console with any VoIP communication software (eg Skype). In the VoIP software is assigned as the audio source (mic) USB device containing PGM signal. As output device USB device module 2602 (USB-1 or USB-2 as you want to use channel A or B module) is assigned.

In order to the channel operates in mix-minus mode jumpers must be placed in the circuit board (next to RJ45 connectors). The following figure shows the location of the jumpers. To mix-minus on Canal-A place jumpers JP1 and JP2. To mix-minus on Canal-B place jumpers JP3 and JP4.



ABOUT USB DETECTION

Before connecting the USB cable to the computer, make sure that both the console and the computer have an effective ground by power cords. For safety, connect a tester in the range of 25VAC between the chassis of the PC and console and verify that the voltage is zero. Only then connect the USB. If there are differences of tension, the USB port on the console or the computer may result damaged.

We recommend do not change the USB cable to other USB ports, to avoid that Windows change the order of USB devices.

Windows 7: Verify that the audio recording device was properly recognized. If Windows 7 recognized it as "microphone device", the recordings will be mono (the same signal in both channels). To correct this: Control Panel → Sound → Record → choose the USB device (shown as USB microphone) and click [Properties]. Then select the tab "Advanced", display the menu of formatting options and choose a format for stereo recording (2 channels, 16 bits, 44100Hz).

2.3.6 Disparo de dispositivos externos

Each module has a '**Start devices'** (TRS 1/8 ") connector to trigger external devices. This signal can command devices remotely from the console when an attenuator is activated; eg Audicom computer and audio processors.

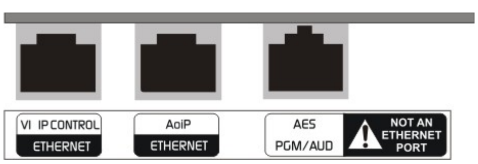
Start Devices' is an output type **"open collector"**. Attenuator with closed and / or the channel off, present an open circuit output. When the channel is enabled (the fader up and 'AIR' button pressed) the transistor conducts, closing the circuit. It can handle up to + 24V / 100mA.

TRS 1/8" START DEVICES		
Punta Canal "B"	**Manga:** GND	**Anillo** Canal "A" (2612 canal C)

2.3.6.1 Uses of the Start Devices outputs

Some radios use an indicator light for each microphone (tally). In that case, the output "External Devices" is used to turn each light microphone; as shown in the general connection diagram (see 2.3.9 - General connection diagram "). "Start Devices" can also be used to launch audio files; or to switch video cameras using the Solidyne Audicom software (requires GPIO-USB adapter); or to trigger recorders, relays or other devices.

2.3.7 Opción VI/Control / AoIP / AES-3



2.3.7.1 AoIP

Please see "2.7 Models 2600/AoIP

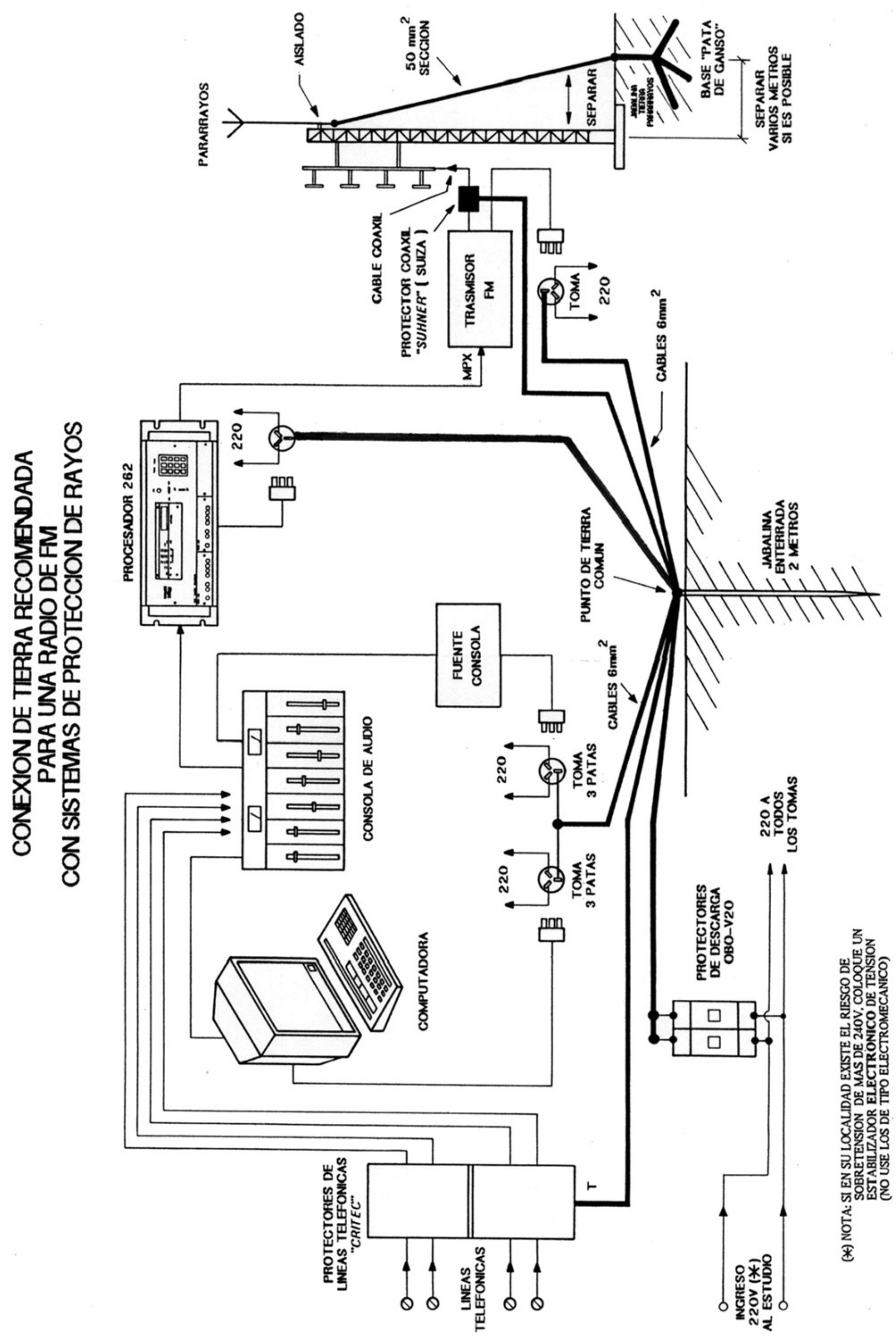
2.3.7.2 AES-3

AES-3 output gives PROGRAM and AUDITION signals. Are balanced with transformer. Uses an RJ45 connector and must be connected using Category 5 STP cable. Consult your dealer for RJ45 cable two male XLR.

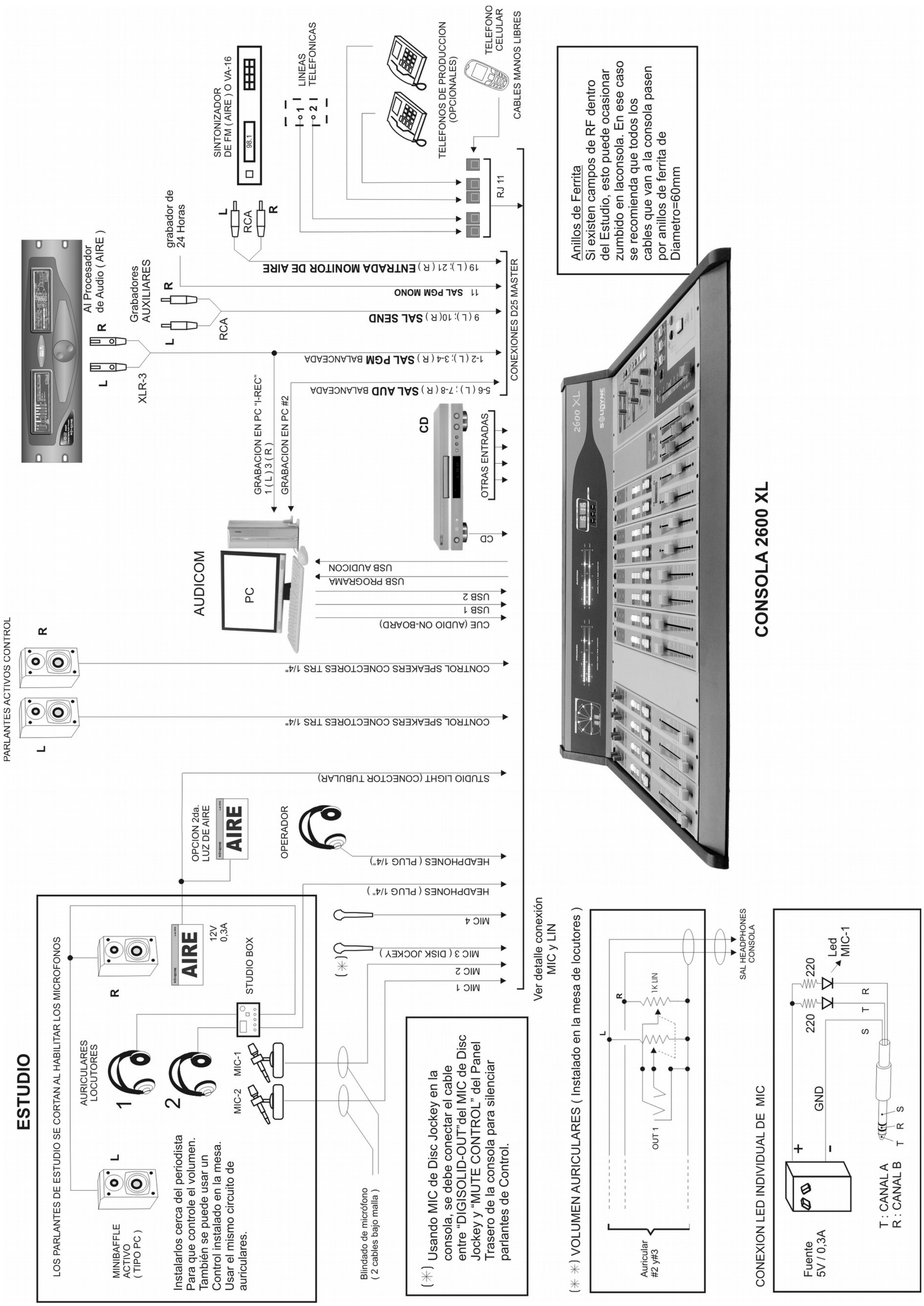
If the AES module is acquired after buying the console, it should be mounted on the Master module by following the directions provided with the AES module.

RJ45			
1	PGM (+)	**6**	AUD (-)
2	PGM (-)	**7**	NC
3	AUD (+)	**8**	NC
4	GND	**9**	Shield (RJ45 chassis)
5	GND		

2.3.8 Ground connections recommended for protection against electric storms



2.3.9 Schematic Diagram – Recommended connection of 2600 consoles



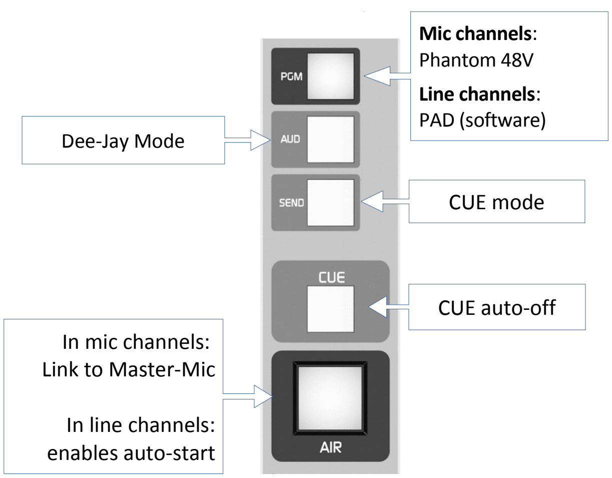
2.4 Customizing the modules

2.4.1 Configuration mode

According to the connections and use, some console's modules can require a customized configuration. This configuration is programmed into each module as follows:

1. Switch off all assignments and close the faders A and B in the module.
2. Press and hold any of the buttons AIR. After 5 seconds the module enters in mode "setup" and several buttons start flashing.
3. Each button enables (on) or disables (off) one function (see 2.4.2).
4. To confirm the changes, up and down any fader of this module. The module flashes and returns to be operating (with all the buttons off).

2.4.2 Customizable features



2.4.2.1 48V Phantom power

Built-in 48V phantom powering is standard; to operate with condenser microphones. When a 2600's MIC module connects to the phantom voltage, the voltage is applied to all channels of these module. Conventional dynamic microphones can be connected without problems to a phantom powered channel, since this kind of microphones are designed to decouple the DC.

2.4.2.2 Using a mic into the Control Room

There are two features that must be changed to be able to use a microphone at the Control Room:

a) Unlink the channel from the "Master Mic" button.
b) Enable the DeeJay mode.

By default, all microphone modules are switched-on when master MIC button is on-air, but this feature can be disabled for each channel. That's very useful for Dj's and showman's who talks on-live from the control room. In this case, the microphone located in the control room will not be on-air by the Master On-Air button.

> ✍ Radio stations in which studio and control are the same room can connect the monitors directly to "Studio speakers" output.

In DeeJay mode, that is, when the speaker operates itself the mixer console and talks from the Control Room; **the Control Room speakers monitors are muted** when this mic is on-air, to avoid feedback; and "auto-cue" feature (that allows *cueing* the microphones after pressing the Talkback button) is disabled to that channel.

2.4.2.3 CUE mode

CUE butons can operate in two ways:

"SOLO" mode: [Set: SEND buton ON (default) enables the SOLO mode for this channel] All the buttons working in mode "CUE-SOLO" are linked. Only one CUE-SOLO can be active at the same time. If another CUE-SOLO buton is pressed it will turn off a previous CUE-SOLO; but will not take effect over other channels which working in CUE-standard mode. If all channels are set as SOLO, only one CUE can be active in the console.

'Standard' mode: [Set: SEND button switched off] When the CUE-SOLO mode is disabled the CUE works in the traditional way: each CUE button is switched on/off with independence from others. When more than one CUE button are pressed, they heard mixed.

A widely used combination is: the microphone channels in CUE-standard mode; and line-level channels CUE-SOLO mode.

CUE Auto-off: [Set: is enabled with the button CUE switched ON (default)] Regardless of the CUE mode used (alone or standard) the auto-off feature turns the CUE when the channel is on the air (ON AIR button on and fader open). CUE can be turned on again while the channel is on the air (it will shut down again if the fader is moved).

2.4.2.4 Link to 'Master Mic' (mic channels only)

Link the channel to the "Master Mic" button, located in the 2607 Master Module. By default all mic channels are linked to "Master Mic". If this feature disables, the channel only can be switched from its own AIR button.

2.4.2.5 Auto-On (only for online channels)

The channel born on when turn on the console (disabled by default). This feature is used in automated stations which are reactivated automatically after a power outage.

2.4.2.6 Remote dimming (PAD)

If PGM button is lighted, that channel is enabled to respond to global dimming 'PAD', available on 'Virtual2600' software. Enabling 'PAD' in the soft, all line channels that have this feature enabled will attenuate by 10 dB.

2.4.3.1 Jumper's table for 2610, 2612 y 2601

MOD.	JUMPER	DESCRIPTION	STATE	CHANNEL
2610	**JE'a** **JE'b**	Bypass **EQ**	**Jumper 1-2 =** EQ disable (by-pass). Jumper 2-3 = EQ enable	See last letter of jumper name

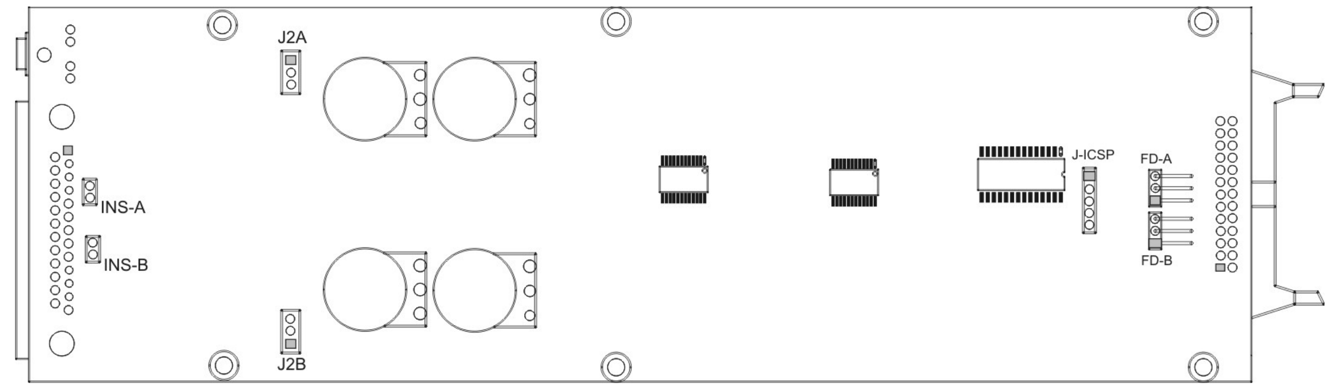


Fig.12 – Mic module 2610 (circuit board side)

2.4.4 Jumpers on Master Module

2.4.4.1 Hybrid's priority circuit

The circuit "priority" of the hybrid attenuates the audio coming from the telephone line when the studio speaker talks. This way to give priority to the speaker in a debate (when both speak simultaneously, the speaker is listened on whom calls) and to improve the audio quality of the local voice. In some uses, like sport transmissions, this effect is undesired (it is not desired to attenuate the atmosphere of the stage when the speaker from studies reads an announcement). The priority can be disabled quitting the jumper **JPRIORITY** located in Master circuit board.

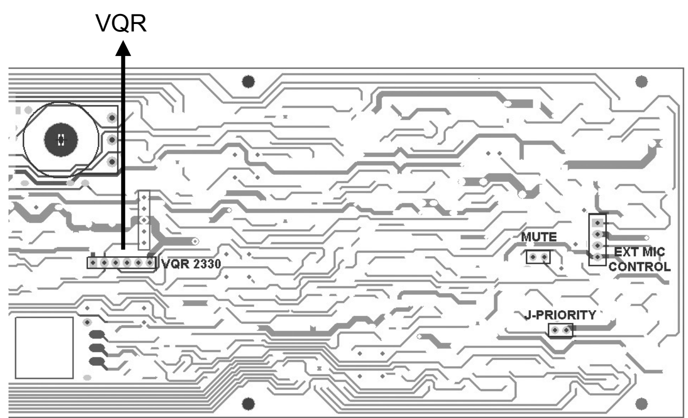


Fig. 15 - Master 2607 – circuit board view

2.4.4.2 External control of Master MIC button

Connector **EXT MIC CONTROL** allows to turn-on remotely the "Masters MIC" button. By the other hand, sends a positive voltage when Master button is ON.

"Master MIC" can be remotely switched joining terminals 3 and 4 of this connector (for example using a relay). When the Masters MIC button is on, there are 15 VCC in pin 1 of the connector.

1	OUT 15 V (Master Mic pressed)
2	GND
3	MASTER MIC ON (joint to turn-on)
4	

2.5 Adding modules to console

2.5.1 USB 2602

The USB 2602 digital module requires connection to the 2607 Master module, to enable the recording on the PC of Audition and Program signals. When you buy this module separately, you receive a ribbon cable with 10-way connector pins.

> If you do not connect the ribbon cable to the Master, the 2602 module works well, but only as playing device. When connects the USB cable, the module is detected by Windows and the USB devices will be available, but if you try to record audio, the computer will record silence.

At the module

The fig.14 (see 2.4.2 – Jumpers on the modules...) shows the 2602 module circuit board side. The connector labeled 'JD' must beconnected to the Master with the ribbon cable supplied with the module.

When you screw the module, be sure to attach the lock washers (star washers) in the top screws of the module, as these washers ensure electrical continuity between the plate of the module and the console cabinet. You can use the washers removed from the blind panel or replace new washers.

At the 2607 Master

You must remove the Master module to connect the ribbon cable; but you don't need to unplug the Master. Locate the 10-pin connector 'JD' and plug in the ribbon cable.

When replacing the Master module into her position, be sure to attach the lock washers (star washers) in the top screws of the module. These washers ensure electrical continuity between the plate of the module and the console cabinet.

2.5.2 2630 VQR module

This optional module adds to the console a processing stage for improve the audio quality of telephonic communications. If you acquire this optional module after purchase the console, proceed as follow to mount it into the console.

The 2630 VQR requires one simple-module space available in the mainframe (it has the same wide of a 2610 module)

Since it requires connection to the 2607 Master module, must be placed beside the Master.

If your console does not have free space next to the Master, considers moving all the modules that are necessary to leave a place beside the Master. Is easy to do it since the modules are interconnected by a ribbon cable with polarized connectors. In order to retire a module of the cabinet, remove the subjection screws.

> Modules and blind panels have a washer type star in the superior screws that assures electric contact between the module and the body of the console; in order to guarantee a correct grounding. Does not omit to return to place these washers after move a module.

Once available the space for the 2630 module you need to retire the Master, to connect the cable that will join it to the 2630 VQR. This cable provides with the module. The previous figure shows the location of connector VQR in the **rear side of the Master.** This is a 6-pins polarized connector, so that the cable can be plugged only in one way.

The module 2630 VQR has identical connector, to which the other end of the VQR cable is connected. This is the only connections that the module requires. The 2630 VQR DON NOT connects to the main bus like others modules.

Place the fixation screws and power on the console. By pressing the on/off button on the 2630 VQR module, this stage is enabled and the NOISE CONTROL indicator lights (it trigger due to the absence of signal from the phone line).

2.5.3 Solidyne “Studio Box”

StudioBox is an optional accessory which groups in a single box to all signals that are needed into the Studio. It offers the following features:

- **5 headphone's** outputs with independent level control.
- Output for **Studio Monitors** with level control.
- **Tally light**
- **Timer/Clock**
- **Talk-back**, to talk with the Control Room operator.

Usually the Studio Box is placed on the table into the Studio; but depending on Studio architecture, a good choice is fix the Studio Box to the wall when the table is next to the Control-Studio window.

2.5.3.1 Connections

The StudioBox connects to the Solidyne 2600 mixer console using a shielded twisted pair CAT-5, with an RJ45 T568B connector (5 meters length cable is included with the console). Through this cable, the console sends to the Studio the signals for headphones and loudspeakers

At the Studios, the **headphone outputs** uses ¼" stereo jack plug (TRS). Any combination of impedance can be used (16; 32 and 64 ohms).

Loudspeakers output are for the Studio monitors, and use a 1/8" stereo TRS (minijack). This is a **line level** output, so powered speakers are needed.

2.5.3.2 Using the Studio Box

About Studio Headphones

Although each headphone has its control of level; the maximum level reached is determined at the console. As you can suppose, what's listening also depends on the selection done in console.

About Studio Monitors

Like happens with the headphones, the loudspeakers output has its own level control; but the maximum level and what you hear depend on the set done on the console.

About Studio Talkback

The white button has two functions:

1. When the microphones are off-air, press the button to speak to the Control Room. The StudioBox has a built-in microphone for this purpose.

   At Control Room, the operator will hear directly by the CUE speaker. The volume is controlled with the trim "mic gain" located at the StudioBox. The knob "CUE level" of the la consola 2600 don't take action over the StudioBox talkbak signal. To answer, the operator at the Control Room will use the standard talk-back circuit of the console.
2. When the microphones are on-the-air, the white button illuminates to indicate that microphones are "on-air". Obviously, in this condition the talk-back feature is disabled.

> The trim located below the MIC allows to adjust the microphone gain, so it adjust the listening level on the Control Room.

About Timer/Clock

The display shows the current time (when the mics are off-air) or the time lapsed on air (when microphones are on). This function operates like the accessory "TIMER" for 2600/TM consoles. Please see 1.4.3 – Timer / Clock for more details.

In the StudioBox, the "counter mode" can be disabled by removing an internal jumper. For make this, open the back cover of the Studio Box and remove the jumper (there is only one jumper). This will disable the lapsed time on-air mode, being always the current time on display.

2.6 Set the inputs gain

These adjustments are necessary to calibrate the input module gain, so that equal positions of faders represent equal outputs levels in all modules. In order to set the input gain for each channel, use a sinusoidal tone of 1KHz@0dB or a calibration CD.

To adjust the input gain, play the test tone and calibrate the presets with the screwdriver provided with the console. Left and Right presets at the front panel, must be calibrated to obtain a measurement of 0VU, with the faders at the center of the gray zone (-15 dB).

To calibrate the PC channel, proceed in the same way playing loud music or voice from the PC. Proceed the same way to calibrate other sources.

REMEMBER

- The audio equipment handle different signal levels: The professional audio level, with balanced outputs operates at +4dBu or + 8 dBu, whereas home equipment with unbalanced outputs manages -10dBu levels.
- **0 VU** refers to the nominal output level. When the VU meter reaches 0 VU, the output level is +4dBu.
- Use the balanced inputs of the console for professional devices, and the AUX unbalanced inputs for home quality devices.
- USB channels don't have input gain trims. The input level manages from the computer.

2.7 Models 2600 /IP

These models include RJ-45 output that sends streaming. It allows:

- Link the console with the transmitter plant. At the transmitter plant the streaming receives using a dedicated hardware (Solidyne ADA102 or Solidyne 562dsp audio processor).
- Link with another 2600 /IP
- Send the program signal to a computer on the network, using a software to receive the streaming.

2.7.1 STL link

In studies, the 2600 /IP console works as encoder to establish a bidirectional link (full-duplex) between Studios and Transmitter Plant. In the transmitter plant, the streaming receives using a hardware like Solidyne ADA102 or 562dsp audio processor.

2.7.1.1 Set the IP

The setings options are accessed using any web browser. By default the unit uses "Dynamic IP", so that when connected to a LAN, gets an IP address via DHCP (the router assigns an IP). The procedure is as follows:

Step 1

Connect the mixer to the network via a standard cable. The network must have a router, so that the mixer obtains an IP address via DHCP. The mixer can also connect directly to a modem-router, as it usually also assign an IP via DHCP.

Once the console gets the IP, is ready to start working. The green LED on the rear panel (RJ45) is blinking.

Can not find a DHCP server then the 2600/IP scans the network for a free IP address (this can take a few minutes).

Step 2

To know the IP assigned to the console, the user must run the application “Discovery_AoIP”, which is available at the following link:

solidynepro.com/DW/IP.exe

The file IP.exe is a self-extracting ZIP. When the user runs this file, a folder “Solidyne IP Discovery” will copy to the HD. This folder contains the apps and instructions needed to obtain the 2600's IP address. Look for leame-readme.txt and follow the indications according the case.

Step 3

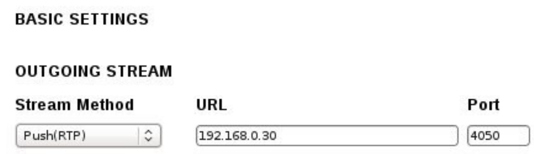
Open a web browser (eg Firefox, Internet Explorer) and enter the IP address announced. The Control Panel of 2600 will appears.

2.7.1.2 Define the destination IP

The status screen indicates the IP port configuration. The module is factory-configured as Studio Encoder (option "Location" → 'Studio Encoder').

Step 4

To check the configuration, access the "Configuration" and choose **"Basic settings"** in the left menu .



1. "Stream method" should be "Push (RTP)"
2. In the URL field defines the DNS name or IP address, and the destination port **to which the console sends the streaming.**
3. Press "Apply" to confirm the settings.

The configurations listed below require knowledge in network administration.

The destination IP address is the external address of the network in the Transmitter Plant, where it is connected DECODER (static IP assigned by your ISP). When the data packets reach the router / firewall on the other end, should be re-directed to the DECODER IP (eg 192.168.0.30).

As the 2600/IP transmitted to a specific IP address and port, in transmission plant all packets arriving at that port on the router address should be forwarded to the DECODER, or a particular computer, which will turn into audio. Identify which packets should be sent using port forwarding.

2.7.1.3 Audio settings

Step 5

Go to Config → Audio

Input source: Default is *Line Stereo.* Don't change it.

Format: Streaming format. Default values are:

Format: **PCM 16 bits stereo**
Sample rate: **48 KHz**

This settings (PCM16 @ 48KHz) generates a streaming of 1.6 Mbit/s.

2.7.2 Using a microwave digital link

A 2600/IP console connected at the studio to a broadband Internet can cover any distance from the studios and the transmitter plant.

For short distances an option is point to point RF link transmiting uncompressed audio (PCM 16 bit/48 KHz). It uses a microwave link for 5.8 GHz (or 2.4 GHz in some countries) using the standard 802.11.x. This band is available in all countries and does not require special authorization. It can cover up to 27 miles if there are no obstacles between the extremes. Logically encoded audio can be transported and for special applications supports bidirectional connection.

For more details, please contact us at info@solidynepro.com.

2.7.3 Decoding using a computer

To receive the streaming from 2600/IP using a computer, set the audio format (see 2.7.1.3 – Audio settings) as **PCM16 @ 44.1KHz;** or any MP3 mode.

The audio player software should supports the RTP protocol. We recommends using VLC Player (http://www.videolan.org/vlc/).

To start to play in VLC, go to *Media –> Open network*

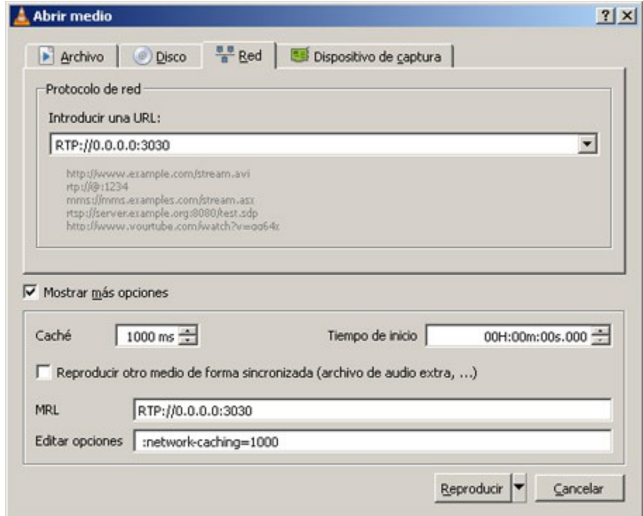
In URL enter:

[protocol]://[sourceIP]:[port]

Example:

RTP://0.0.0.0:3030

In where RTP is the protocol used by the Solidyne 2600/IP; 0.0.0.0 means *from any incoming IP* and '3030' is the port used. *Remember to enable your computer firewall to allow incoming streaming trought port 3030 (or the port that is used).*



NOTE: *Some versions of 'VLC Player' also supports the string* RTP://@:3030

Using the mixer — Chapter 3

3.1 Introduction

You can discriminate different areas in the console: the input channels; the monitoring levels control, phone hybrid and talk-back (master 2607); and the VU-meters panel.

The audio signals coming from the PC, Satellite, DAT, Mini-disc, Microphones, etc, enters to the console through the input channels that amplify them. You can control all sources levels using the main faders, and to listen the signals before send to the air pressing the CUE buttons.

To send a channel on the air, press the AIR button and open the main fader until reach the desired level. The buses assignments button’s (PGM, AUD, and SND) sends the channel’s signals to the Master outs.

Each input section can receive signal of two different sources: Microphones or stereo Line. The source is selected by a switch located at top of the module. Remember that the modules are doubles: each module has two faders that manage two inputs. Depending of the model this inputs can be:

- 2610 Microphone module manages 2 microphones (MIC) and 2 stereo unbalanced inputs (AUX).
- 2612 Microphone module manages 3 microphones with EQ and processing, and 3 stereo auxiliary inputs.
- 2601 Line modules manages 2 balanced stereo line inputs (LIN) and 2 unbalanced stereo (AUX.)
- 2602 modules manage two USB digital inputs and two analog auxiliary inputs

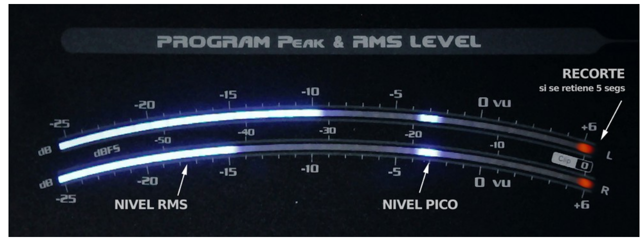
The 2607 master module manages the monitoring circuits, Talk-back, and telephone lines.

The turret contains the level meters that show the program level; and a speaker for CUE.

3.2 Meter bridge

3.2.1 Audio meters

The program level meter has a **dual scale**: average level VU and peak level dBfs. The floating LED displays the signal peak.



The last scale LED (red) has a dual function: if occasionally ligths, indicates +5 VU in the scale of average level. But if the level drops and the last LED is retained (5 seconds), it indicates CLIPPING. The fact that the level meter reaches the full scale does not always mean that digital clipping occurs; only is clipping if the last LED is retained. Take in mind that when Vumeter is at full scale, dBfs peak reading is lost because the peaks are "covered" by the average level indication.

The headroom is 15 dB (0 VU = -15 dBfs as AES recommendation K15).

The 2600XL models also features level indicator for AUDITION bus while 2600XD models incorporate meters for the SEND bus.

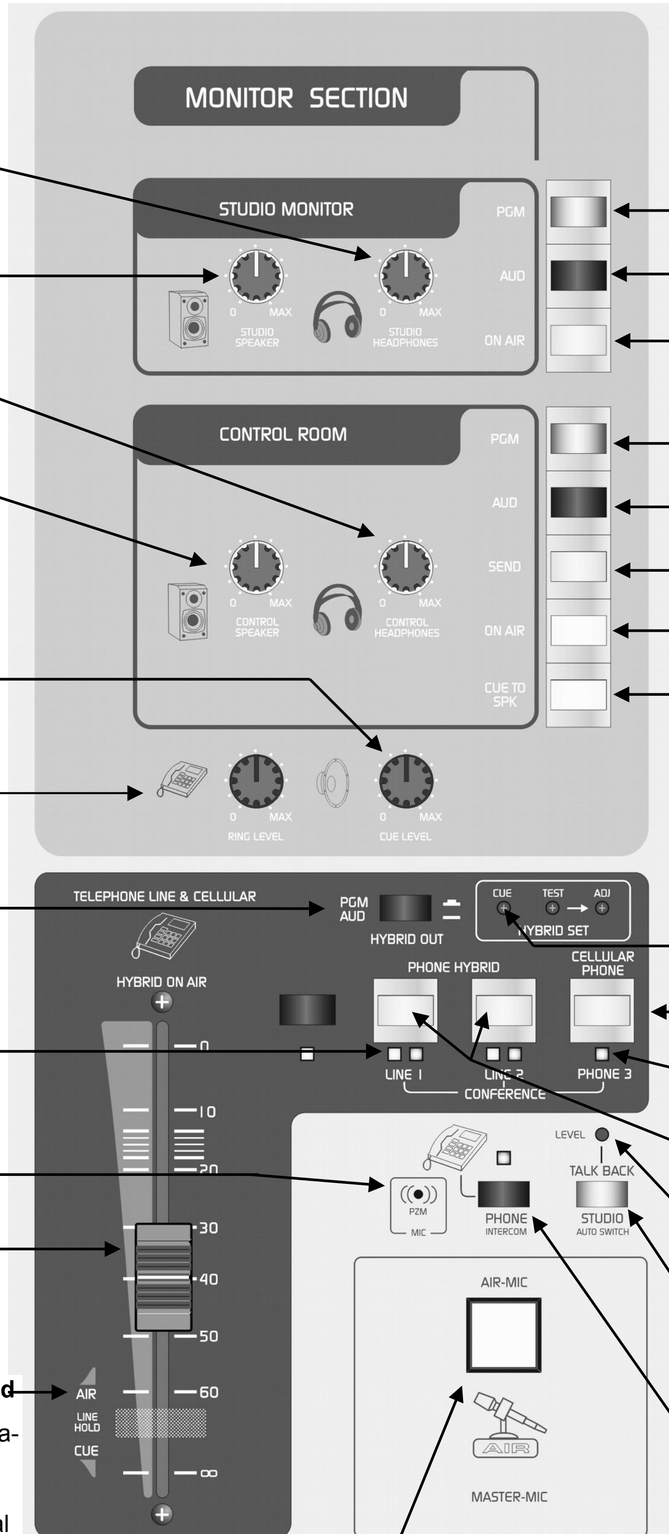
3.2.2 Clock / Timer - Set the hour

To adjust the time there are three buttons: MODE, UP and DOWN.

- Pressing MODE button the adjustment starts. The less significant minute will blink. With Up and Down buttons, change the value.
- Pressing Mode again you can change the most significant minute.
- Finally, pressing Mode again the hour character will blink. Set the hour and press Mode to finish the time adjustment.

MODELS XD: When a console model XD connects for first time to the software 'Virtual 2600'; the console take the current time from the computer.

3.3 2007 master module



Studio headphones level

Studio monitors speakers level

Control Room headphones

Control Room monitor speakers

CUE level. Manages the level of the built-in speaker

Ring level (it's listened on the CUE speaker)

Hybrid Output. **PGM** to send on-air **AUD** to recordings.

Line calling LED

PZM MIC for **Talk-Back** and telephonic communications

On-air hybrid's level. Automatic switching between CUE - ON-AIR.

Automatic switching threshold

Cue: establish the communication by the CUE speaker (or free hands mode).

Line Hold: Sends PGM signal to the caller.

AIR: When the fader is over this indication, the calling is sent on-air.

Assigns **PGM** to Studio

Assigns **AUD** to Studio

Assigns **'AIR' input** to Studio (external tuner).

Assigns **PGM** to Control Room

Assigns **AUD** to Control Room

Assigns **SEND** to Control Room

Assigns **'AIR' input to** Control Room

Assigns **CUE** to Control Room main Monitors

** If several buttons are pressed, the signals are mixed.*

Hybrid's rejection factor (aka Null). (See the following pages)

Adjusts the hybrid level sent to CUE

Takes the audio from **Cell phone**

Line taken LED's

Takes the phone lines. Both lines and the cellular can be in conference.

Talkback level preset

Talkback button

Talk to telephone line. "Free hands" dialog mode. When this button i pressed the Control Room monitors are muted. The calling is listened on headphones. The PGM audio stays in background

Master MIC button. Enables all microphone channels and lights the Tally.

3.3.1 Telephone line Mini-Central

The Master 2607 manages 2 telephone lines and one cell phone, operating as multi-line system. It allows conferences between a journalist and 3 phone interviewees. The four peoples can be on the air, in conference, at the same time.

Cell phone connects to the console using the "free hands" connector (see "2.1.5 Telephone Lines"). The cell phone also supports conference with the lines.

The operation is easy and safe against operation errors, due to its logic of security and automatic control circuits. In spite of being an advanced call center system (and not a simple Hybrid), we will continue denominating it hybrid from now, due to their habitual denomination.

3.3.1.1 Operation

The 2607 Hybrid must be connected as is indicated in the Chapter 2 of this manual. The operation procedures are the following ones:

When a call is incoming; press the line button with the blinking LED. The 'ring' will be listened on the CUE speaker (this volume adjusts with the RING knob).

You can **answer the call** in two ways: **a)** From the telephone set associated to that line or **b)** from the hybrid of the console. To answer from the console, press the blinking button, the hybrid will take the line and a pilot light will indicate that the phone line has been taken by the Hybrid.

The ON AIR Hybrid fader must be closed so that the communication will be routed through the CUE circuit. Under these conditions, you will listen to the caller in the CUE speaker. The level is adjusted using the CUE knob. To dialogue, use the Talk-back microphone, by pressing the button PHONE. This button is retained, so you must press it again to listen the answer at the CUE speaker, since it's a half-duplex communication.

But you can dialog in **full duplex** and **free hands**, staying pressed the button PHONE. In this mode, the Control Room loudspeakers are muted to avoid loop-backs, so you must use headphones to dialog with the caller (when the two lines are taken, the talk-back MIC sent signal to both lines).

REMEMBER

- If you listens high return with boosted basses of your own voice in headphones; the rejection of the hybrid badly is fit. It can directly fit maintaining it the communication in "free hands" mode, and varying preset NULL until finding the point in where the return of your own voice is smaller. This adjustment is equivalent to the explained previously.
- If the communication listens saturated, with high background sound, the level of preset CUE is too high. Diminishes this level until obtaining an optimal listening. With the console a small screwdriver is provided to make these adjustments.

When finishing private conversation, **release the PHONE button** and use HOLD position of the Hybrid ON-AIR command to send the program signal to the telephone line. This mode, named waiting mode, allows sending the program signal to the caller.

Finally, when you open the ON AIR fader, the telephone call is sent to the air, and the audio program returns to the caller so that he can dialogue with the journalist. The CUE return is disconnected automatically.

> Please do not exaggerate with the level on-air of a communication. Due to the analogical hybrids are not perfect, a part of the audio sent to the line, that is the voices of the study, is not cancelled and returns through hybrid, adding itself to the direct voices of the study (this is what fits the rejection). Working with normal levels this effect is not noticeable; but using the hybrid at high levels, the voices of the study can be soiled (coloration).
>
> A telephone communication not necessarily must arrive at 0 VU to equal the loudness of the voices of the study. Have in mind that the LF, that carries greater energy, is not present in the telephone line, but in the voices of the study. The lows are those that produce greater deviation in VU METERS. Nevertheless, for the ear the loudness is defined by the mid rage frequencies. So that if you note that VU Meter "peaks lower" with the telephone line that with the voices of the study, this not necessarily means that "it is listened lower". On the other hand, the audio processor of the radio will be in charge to equal both signals. **Conclusion:** you do not leave the VU deceives to you when mix voices of the floor with telephone calls.

To **retake the communication** in private mode, close ON AIR fader and press the line button. The line will be addressed to the associated telephone set. If the telephone is hanged, the communication ends.

3.3.1.2 HYBRID OUT

This button allows to assign the hybrid's to PGM or AUD bus. Sending the hybrid to AUD you can record a communication without it leaves on-air (one line channel assigned to PGM with music; MIC's, hybrid and monitors assigned to Audition). Note that when hybrid is assigned to AUD, it only receives signal from channels assigned to AUD buses.

3.3.1.3 Return to phone line

Most of the existing telephone hybrids on the market today, were designed over 30 years ago for analogue telephone exchanges (PBX) Solidyne hybrids, however, have been recently designed for private or public telephone exchanges today, which are fully digital. The new technology Hybrids are recognized because they have no control of air return level to phone line. This is because inside the hybrid Solidyne uses an audio processor for return signal that includes AGC, peak limiter & audio signal filtering. Therefore the return is auto-

matically adjusted during the transmission and its level is the maximum allowed by the modern digital telephone exchanges.

If you want to check the return level to phone line, you must use an oscilloscope to be placed in parallel with the telephone line and must verify that the signal is 2 volts peak to peak.

Please note that above this level the return can produce problems that will cause intermodulation distortion in the audio signal that goes to air. So in Solidyne hybrids we use a processed return channel, to avoid distortion at the on-air signal. There are hybrids manufacturers that maintain the return control level as they did in the past. This allow operators to adjust "by hunch" this critic level. This makes the voices of the reporters and interviewed people distorted or with coloration.

In Solidyne we keep a high grade of excellence in the audio quality of the hybrid on-air sound. And that quality do not depend on the operator settings.

Note that the Solidyne Hybrid on- air audio quality of the local journalists is ever perfect and without any coloration.

To achieve this level of quality we use a narrow-band

return filter. Then the return signal is limited to the band 400 – 2.200 Hz in order not to distort the signal to the air. This narrow band intelligibility remains high (due to processing) but occasionally may seem to the remote people that it "has little volume" because his band is narrow. This should not worry because it is a subjective sensation that does not affect the intelligibility of speech.

3.3.1.4 Using a cell phone

The work-flow is the same one that used for standard phone lines. The difference is that the call is not taken from the console, but from the cellular phone (you don’t need to press the line buttons in the console). When you answer the call from the mobile, the audio enters to the console through "ON AIR" fader. We can listen who calls in previous and engaging in a dialog pressing PHONE (CUE); to leave the call in WAITING (AUDIO SEND) or to send the call to the air (AIR).

If you need to call from the cell phone, you can make the call and next to plug the hands-free cable to the cell-phone. Or you can make the calling with the cell-phone connected to the console, hearing by CUE speaker and pressing PHONE to dialog (hybrid fader in CUE position).

3.3.1.5 Hybrid rejection adjustment (NULL)

The rejection factor expresses the capacity of the hybrid to avoid that the transmitted signal returns to the system. Whichever greater it is the rejection factor “more clean” will be the audio quality of the studio voices. In order to fit the rejection, proceed as follow:

a) Establish a phone call through the hybrid. Listen it in the CUE speaker, closing the ON AIR fader. **b)** Press and hold the TEST button using a pencil **c)** adjust the preset ADJ until minimizing the audio level (PGM) that you hear on the CUE speaker. Increase the CUE level, if needed, to hear well this minimum level.

Another method is to adjust the rejection factor while a calling is answered in free hands mode, turning the preset ADJ until obtain the minimum level of your own voice at the headphones (see 3.2.1.1).

3.3.1.6 Make a conference on the Air

The conference must be done previously, calling from the associated telephone sets and sending them to the air by pressing the line 1 and line 2 buttons. Let us suppose that the speaker is dialoguing with an interviewee that called (or it was called) by the line number 1; and we want to add to the chat to another person. We make the call from the line number 2, with the associated telephone set. Once the communication has been established, we press the Line 2 button to send it to the air.

CONFERENCES between CELL PHONE and traditional LINES are SUPPORTED.

3.3.2 Monitors and headphones

The Master 2607 monitoring section has two sections: “Studio Monitor” y “Control Room”. Here you find the monitoring selection (what signal do you hear) and the volume control for monitors and headphones of Studio and Control Room.

Monitor options at the Studio are:

- **PGM:** to listen directly the main output of the console (the signal that is sent to the transmitter).
- **AUD:** Allows you to hear only the channels assigned to AUD bus. This way, assigning only a channel to the AUD bus, you can hear the audio of that channel in the main monitors and headphones, while the console remains on the air. Usually this bus output is connected to the recording system.
- **AIR:** Switch to an external output, typically connected to an AM/FM receiver, in order to hear the real transmission of the radio station. This is the recommended method for all radios.

For the Control Room all previous options are available, in addition of the following ones:

- **SND:** Allows hearing only the channels that are assigned to the SEND output.
- **CUE to SPK** (cue to speaker): Allows to send the CUE signal to main monitors. When this button is pressed, being CUE pressend on one or more channels, the "CUE" will be listened by the Control Room main monitors and headphones, while the PGM signal remains attenuated in background.

  Although this button remains pressed, it will not produce no effect if does not have audio signal in CUE. "CUE to Speaker" acts by audio detection on the CUE bus, so that it only attenuates Program when audio is present on CUE bus.

REMEMBER

Although the final level regulates from the loudspeakers and headphones level knobs; the level with which CUE is sent to monitoring is fixed, that is to say, it depends on the own input signal. By such reason CUE can sound higher than the PGM signal. If you are monitoring at a high level, agree that you lower the level of the loudspeakers (or headphones) before pressing "CUE to SPK" (or CUE if "CUE to SPK" is already pressed).

Into the Studio we recommend to install a headphone distributor; like the Solidyne Studio Box. Studio Box brings 5 headphones outputs with independent control levels.

Remember that when microphones are on-the-air; Studio monitors are muted, to avoid feedbacks.

3.3.3 CUE

The unit has a built-in loudspeaker for previous listening. Each channel has a button named CUE, that allows to listen the signal present in that channel leaving off-air (button "AIR" off and main fader closed). If you press CUE in several channels, the CUE signals are mixed. At the Masters, CUE knobs manages the level of previous listening in loudspeakers or headphones according to correspond.

CUE signal can be sent to the Control Room main monitors (and headphones), as was explained previously.

3.3.4 Master MIC button

The **MASTER MIC** button (located at the 2607 Master module) activates all microphone modules, and mutes the Studio monitors to avoid signal feedback. Obviously, the headphones are not muted.

MIC channels can be customized (by jumper) to disconnect them from the Master MIC button. This way, the channel only can be activated from its own AIR button.

3.3.5 Talkback

The talkback system allows the operator to dialogue with the speaker, located into the studio. The talkback controls are located at the 2607 Master module. Solidyne uses a proprietary AutoSwitch system

To make a communication, press the **Studio** button; the Control Room monitors will be muted to avoid signal feedbacks and the operator will be listened into the Studio at the loudspeakers. The Program is attenuated staying in background. This is made for not losing the air program reference. Otherwise, if the order is too long, they will lose the listening to a correspondent that speaks from a stadium or reporting news.

When the talkback button is released, the **AutoSwitch** system activates during 3 seconds CUE on all MIC modules, to listen to the speaker's **answer**. Pressing the TBK button again, this time lengthens. The operator hears the sum of all the microphones. Then, still when the speaker is far from one microphone, be taken by another.

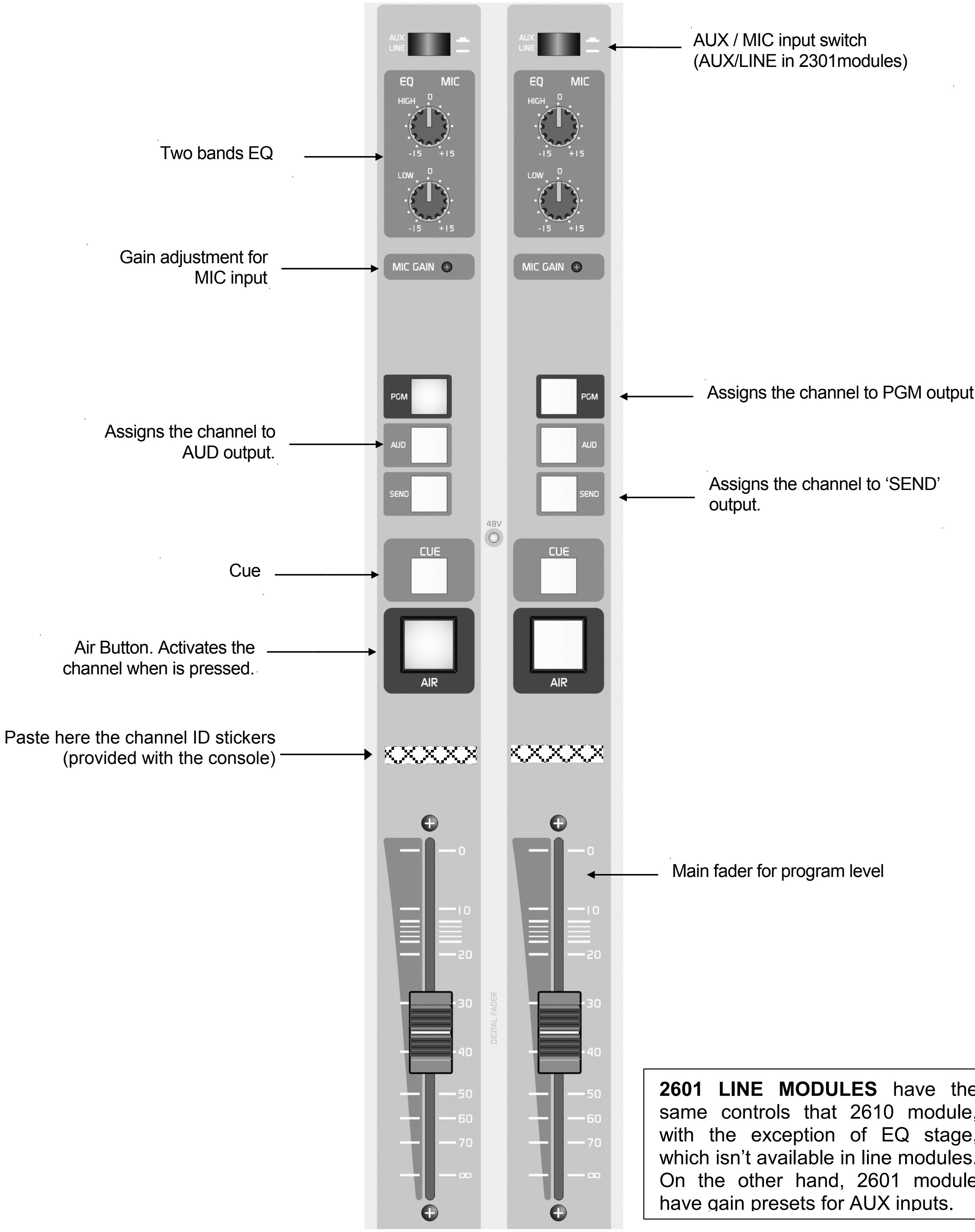
If one of the microphones is used at the control room instead the studio, you must use a microphone with on/off switch. Make sure that the MIC switch will be at **off position**, to avoid acoustic feedback. On modules 2612 you can modify the jumpers to avoid auto-CUE on talkback in that channel.

In this way you get the dialogue by **pressing only one button.**

✍ The level of the talk-back microphone is calibrated in factory, and usually isn't necessary to modify it. If you need to change it, there is a preset located above the Talkback button.

If you have the Solidyne Studio BOX, you can talk at any time with the Control Room (using the Talkback button of Studio BOX); without having to hope that the operator is watching trough the window.

3.4 2610 Microphone module & 2601 Line module



2610 Microphone module

2601 LINE MODULES have the same controls that 2610 module, with the exception of EQ stage, which isn't available in line modules. On the other hand, 2601 module have gain presets for AUX inputs.

3.5 2612 Microphone Modules

At the right, a channel of a **2612** module is showed. The microphone module 2612 has three channels located in the same panel. This module requires the same space that two standard 2610 modules. The EQ and processor works with the MIC input only. The stereo line input has no processing.

AUX/MIC input selection

MIC input gain

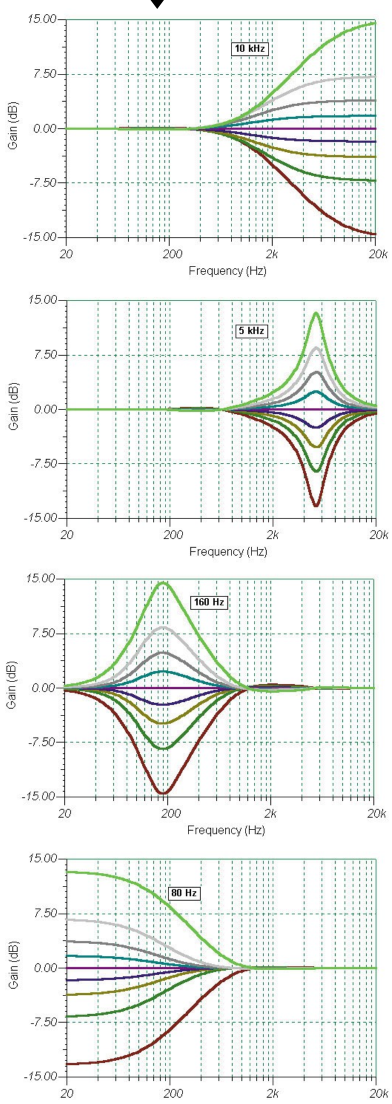
4 bands EQ

Pan-Pot

Gate threshold. Turn the knob to the left to disable the gate. To use the gate, turn the knob slowly to the right, until the background noise is reduced. Listen carefully into the pauses while speaking.

Activates or by-pass the compressor/expander stage.

EQ On / Off



Output buses assignment (PGM, AUD, SEND)

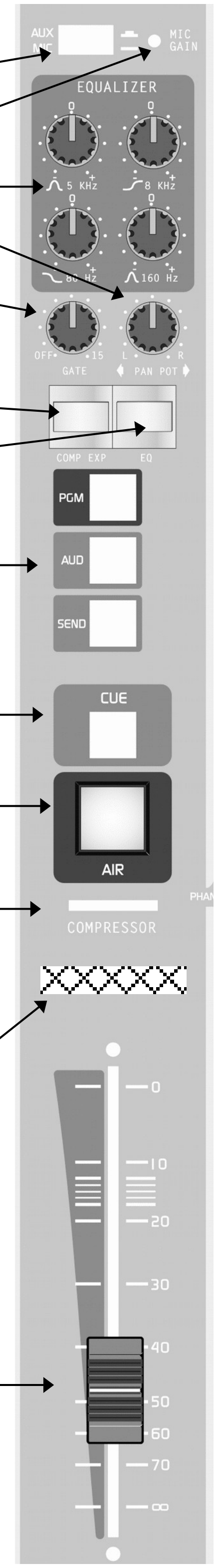
CUE, for previous listening in the built-in speaker.

AIR button. Press this button to send the channel to the air.

Indicates the action of the compressor, which manage the dynamic range to avoid noticeable differences in the voice level. The compression degree regulates with the main fader. Will must operate at the midle of scale.

Paste here the channel ID stickers (provided with the console)

Main fader. Manage the on-the-air mic level.



3.6 Special features of 2602

The 2602 line modules offer two USB connections to connect directly to two computers (see 2.3.1.1 - Connecting the USB 2602 module). There are not operative differences respect the 2601 line modules, but 2602 have an extra feature: the ability to use the computer to make communications using VoIP software (eg Skype).

They also allow the management of digital hybrid conference, as each module generates its own output Mix-Minus. In particular using hybrid Solidyne DH-400; Module 2602 can remotely command them allowing an unlimited number of simultaneous telephone channels.

3.6.1 Operation with Skype

The USB connection can be used to make communications running a VoIP software on the computer (Skype or similar). The VoIP software must confugured as follows:

- In the VoIP software, assign the desired USB channel as the output device.
- In the VoIP software, assigned as input device (microphone) the output of the 2602 channel-A or channel-B (which appears in Windows as USB recording device). See 3.5.1.2 the jumper settings at DB25.

The channel-A and channel-B, with the correspondent jumpes, becomes mix-minus sends. This allos to manage a grat number of channels working with VoIP of telephone lines simultaniusly. Audio that delivers VoIP software directly enters to the console by a 2602 module channel.

3.6.2 Private comunication using VoIP

To talk privately proceed as follows:

- To listen to the caller without be on-the-air, press CUE in the VoIP channel. The audio from the PC will be heard in the CUE speaker.
- To talk, press and hold the CUE button. This enables the talk-back microphone in the 2607 Master. Take in mind that this communication is half-duplex, so you can not listen to the other people while you speaking.
- When you release the CUE button, talk-back mic is muted and the CUE turn on.
- To send the comunication on-the-air, press AIR button and open the main fader.

3.6.3 Jumpers into 2602's connector

In order to enable the private circuit, you need to plug a DB-25 connector in the 2602 module. This connector must joint pins 5-15-17 if VoIP is used over the Channel-A; and pins 6-19-20 if used over the Channel-B.

3.7 2630 VQR Hybrid's processor module

If you acquired this module after purchase the console, please see “Chapter 2 – Installation” to connect it.

Solidyne **VQR**® (Voice Quality Restoration) is a type of audio processing that allows improving the audio quality of a telephone communication. This technique bases on the reconstruction of the spectrum lost due the transmission.

As you know, the bandwidth transmitted through a telephone line reduces approximately to 300 Hz – 3.000 Hz, because this is the range of the human voice. Therefore, the components of low and high frequency, presents in the original signal, are lost in the transmission. These components, although are not important for the understanding of the words, ARE VERY IMPORTANT FOR THE AUDIO QUALITY, because they give “weightiness” and “presence” to the voice. System VQR really reconstructs the bass of the voice, reaching the frequencies of up to 50 Hertz, being able also reconstructing component of high frequency to recreate the highs that are of extreme importance to obtain the *presence sensation*.

In addition, stage VQR has a third control to improve to the dynamic range, obtaining values of up to 70 dBA in a telephone transmission.

This processing is applicable as much to calls by terrestrial lines, like a calls made through the mobile telephony. Although the reconstruction reaches to callings made using the internal microphones of the cellular or the fixed telephones, the **best results are obtained using a portable audio console** and dynamic microphones of good quality.

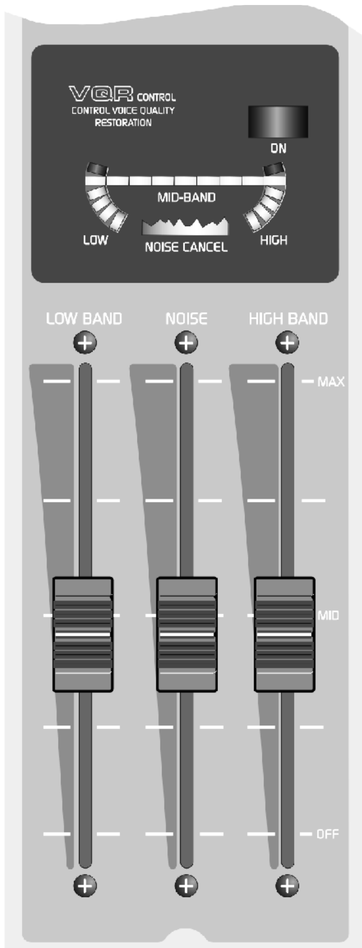
For details on VQR, you can consult the information available in the section *“Technical Documentation”* on our Web site.

3.7.1 Using the VQR

Module 2630 VQR works as much with the internal hybrid of the 2600 console like with any external hybrid; connected to the Master 2607 using the “External Hybrid Send & Return” connections.

The module has a complete panel with luminous level indicators that shows:

- Degree of restoration for low frequencies.
- Degree of restoration for high frequencies.
- Action of the Noise Cancel



The VQR processing can be enabled or bypassed pressing the on/off button on the module. By pressing this button the VQR stage activates and the MID BAND indicator lights, (or NOISE CANCEL if the fader NOISE is open, due the absence of audio signal). With the button up the module is bypassed and the audio of the calling is sent on the air without processing.

The user adjusts the amount of processed signal using the faders "Low band" and Hi Band".

Both controls even have an ample rank of work, making possible the processing in telephone signals whose bandwidth is very restricted.

Low band

Manages the level of lows added to the original signal. With the fader closed there is no reconstruction for low frequencies.

The level of reconstruction, or the amount of low frequencies that is possible to add to the signal, **depends** on the **audio quality of the telephone line** (all communications don't transmit the same bandwidth) and the **telephone or microphone** used at the other end. Obviously, same results are not obtained using the small microphone of a cellular telephone or a microphone of good quality with a portable console. at least quality has the transmission (smaller bandwidth) smaller will be the action of VQR processing.

Make sure to listen to the processing in the main monitors of the control room, to avoid an excessive reinforcement of lows in the processed signal; that can take place if you are monitoring the communication using small headphones or loudspeakers of bad quality.

Hi band

It controls the level of high frequencies added to the original audio coming from the telephone line. With the fader closed the high processing deactivates.

The action of this control **is much more critical that the Low Band**, since an excess of highs processing will generate an *"artificial"* sound; and in extreme case *"crashed high"* sound can take place, that will be annoying to the listener.

On the other hand, consider that an A.M. radio can require more emphasis in high frequency than a FM; to obtain a well-known improvement on the air; therefore the control Hi Band **has an ample rank of action**.

The reconstruction level -or amount of highs added to the signal- depends on the quality of the transmission. This stage will be affected, mainly, if the line has much background noise.

⚠ **REMEMBER**
• The optimal level of work is obtained when the indicator MID BAND lights with the signal peaks. Lower audio levels can affect the behavior of the VQR processing.

NOISE Control (expander)

This control is used to reduce the background noise present in the phone line. It acts only during the *silences* in the conversation, attenuating the level of the signal to suppress the noise. This is

quick action gate reason why its effect is imperceptible with normal levels of noise, not affecting the word.

The NOISE fader acts changing the **threshold** of the expander/gate. When de background noise is under this threshold the expander/gate works attenuating the noise.

Closing the fader the expander/gate is turned off. When opening the fader **increases the threshold**, that is to say, the signal level below which the expander/gate goes off. The action of the expander/gate is showed in the display by the NOISE CONTROL indicator.

How use this control

Increase the threshold raising the NOISE fader until eliminating the background noise. An insufficient level will do that the noise remains, although reduced. An excessive level will cause that the audio appears “intermittent”.

Next some important tips to take in mind when use this control:

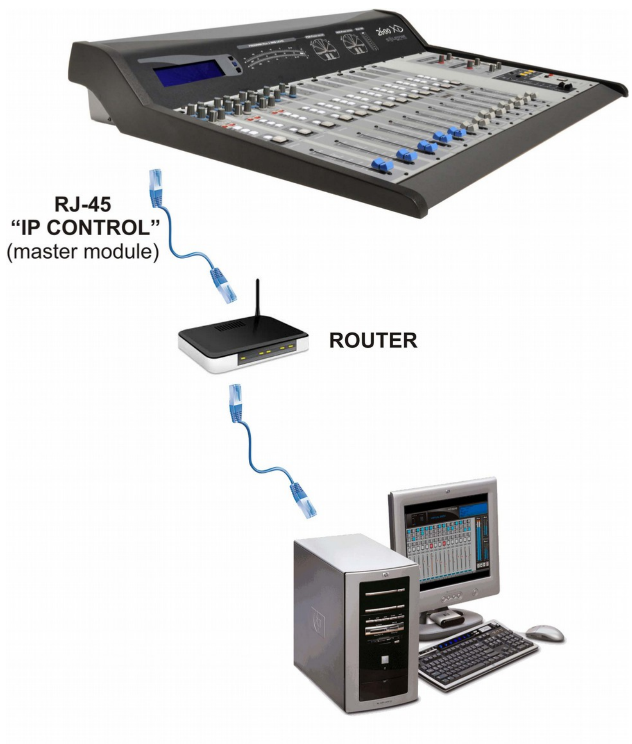
- If the background noise in the communication is very high, will be always over the maximum threshold (fader at top) with which the expander/gate will not work correctly.
- Consider that the expander/gate releases whenever the audio signal is below the threshold. If the background noise is very variable in level (noise from a street, for example), it agrees not to use the NOISE CONTROL to avoid that during the pauses it activates and deactivates generating an intermittent background sound. In these cases it is preferred to leave the ambient noise.
- Also can happen that the background noise is very notorious (a strong humming or buzz) and although the gate can attenuate it during the pauses, the effect “appearance” and “disappearance” of the noise is more annoying than the own noise, due to a psycho-acoustic phenomenon according to which the ear “is accustomed” to the floor of constant noise when concentrating the attention in the word.

According to these advice, the good criterion of the operator will determine when it will make use of the noise gate and in which cases it will prefer not to use it.

3.8 Remote control

Models with Ethernet controller (option /VI) can be managed remotely via LAN, using any web browser on any operating system.

The console connects to a LAN from the RJ45 connector labeled "IP Control", located at rear panel of Master 2607. This connection uses a standard Ethernet cable. By default the console is set to work with DHCP. When the console connects to a router/switch, it will assign an IP address.



Once the 2600 / VI is connected to the router, the user accesses the Virtual Interface Control from any computer on the LAN, using a web browser and typing in the address bar the following:

2600XD/

- If the connection is successful the login screen of the 2600 / VI appears. See "3.8.2 - Virtual Interface" on instructions from the Virtual Control Interface
- If the login screen does not appear:
  a) Check the connection of the console to the router on the LAN and try again.
  b) Use the "Solidyne IP Discovery" tool to find the IP address of the console. Enter the IP address in a browser to access the Control Interface.

3.8.1 Solidyne **IP Discovery**

If due to network settings can not access the console using the generic name "2600XD /"; must be accessed using the IP address. The IP Discovery Tool lets you know the current IP of the console.

Download the following file:

solidynepro.com/DW/IP.exe

The file is a self-extracting ZIP. When run is created in the local folder a sub-folder called "Solidyne IP discovery" that contains the applications and instructions. Look in that folder the "readme.txt" file and follow the steps as required. Once the 2600's IP address is obtained; enter it in a WEB browser to access the "Virtual Interface Control.

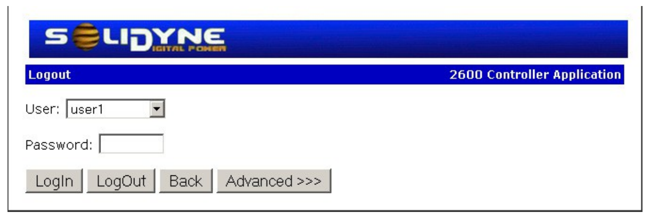
IMPORTANT: The "IP Discovery" tool also must be used when:

- You want to **connect the console directly to a computer** using a crossover cable (no LAN) for operational tests or demonstrations.
- DHCP is disabled and an unknown fixed IP is set to the 2600/VI, being outside the range of the LAN. In this case "IP Discovery" will be used to find the current IP address to access the Control Panel (either to enter a valid IP address or enable DHCP).

For details about direct connection using crossover cable, please refer to the additional documentation included with the IP-Discovery.

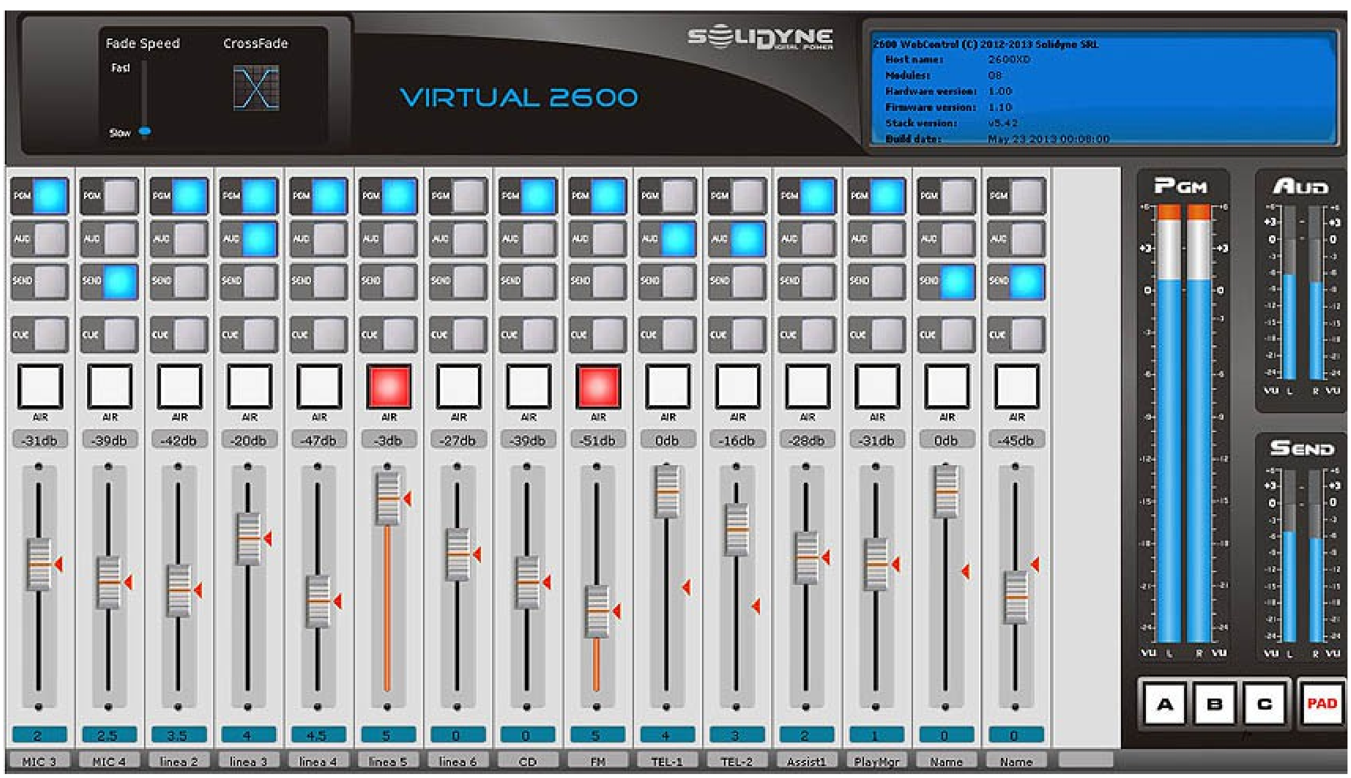
3.8.2 **Accessing to Virtual Interface**

To access to the control panel, **user name** and **password** are required. There are five factory users: user1; user2; user3; user4 and user5; being the default password for all '1234



Click "Login" to access the control interface. The option "Advanced" allows to access the configuration options, which are detailed below.

3.8.3 Virtual Interface (/VI option)



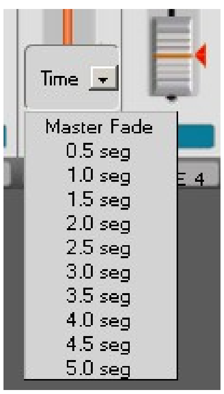
Virtual2600 operation from the software is similar to the operation in real console. The software interface has some extra facilities:

- Automatic crossfades with adjustable time.
- Microphones can group up to three masters buttons.
- "PAD" button attenuation for line channels (assigned by hardware).

3.8.3.1 Manage the levels

In each channel level is set from the Master Fader. The red pointers indicate the position of the physical fader on the console. As the console faders are not motorized, changing the level from software positions between the virtual fader and hardware do not match. **The current level is always determined by the virtual position in the software.** If the actual position is changed in the console, the level is updated in the software and both agree

Turning on / off a channel (AIR button) progressive level change occurs. The speed is determined for each channel, and is indicated below the fader. Clicking on the blue tag a menu is displayed 0.5 times to 5 seconds, at intervals of half a second. The "0" value links the channel speed to the control "Fade Speed".



3.8.3.2 Cross-fades

This function produces a channel attenuation and the progressive appearance of another. The **fade time** is determined from the "Fade Speed" control, varying between 0.5 (Fast) to 5 seconds (Slow).

The image below shows the procedure to perform a crossfade:

3.8.4 Advanced settings

The "Advanced" button to access the options in Virtual 2600. *Username* and *password* is required. Default is 'admin' for both.

3.8.4.1 Users settings

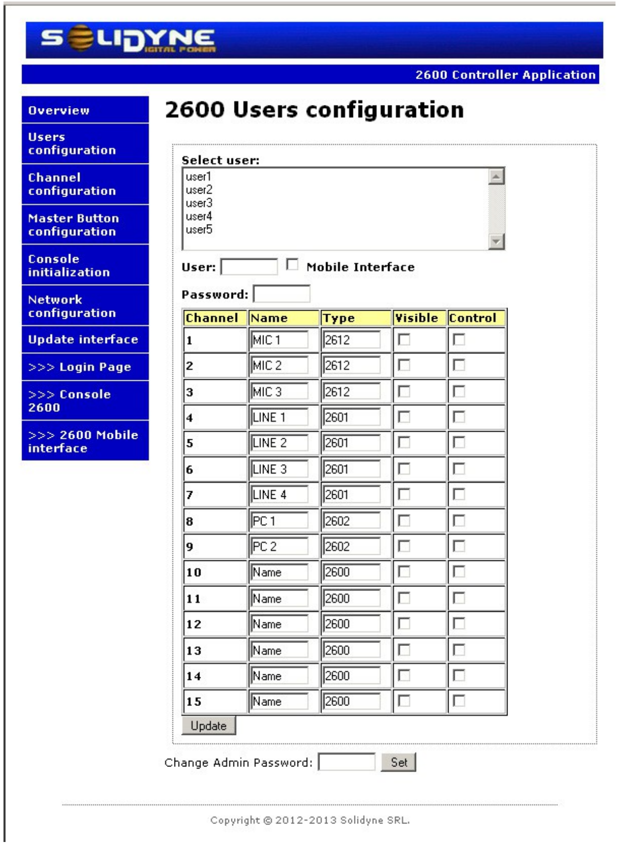
Can be defined up to 5 users. The "User" field allows to update the name for the selected user.

"Mobile Interface" enables a reduced version of the graphical interface, designed for mobile devices. If this option is enabled, this user will always enter the 'mobile' interface (no matter the machine that is being accessed).

"Password" allows entering a new password for the selected user.

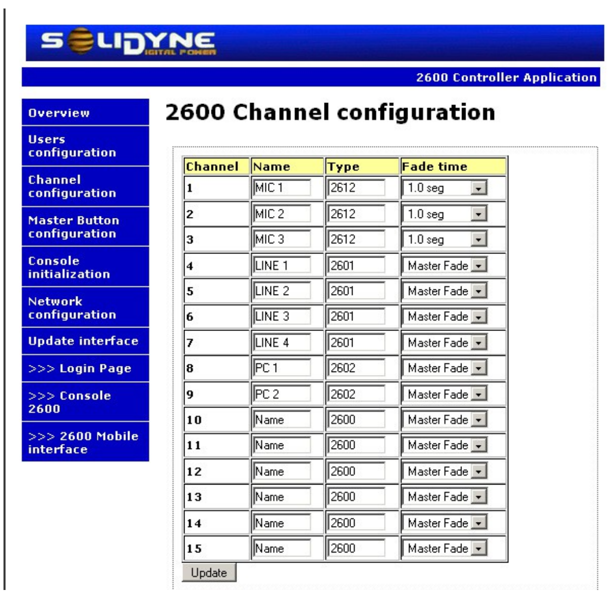
For each user can define the visible channels, and enable / disable the control of channel.

To save the changes click "Update"



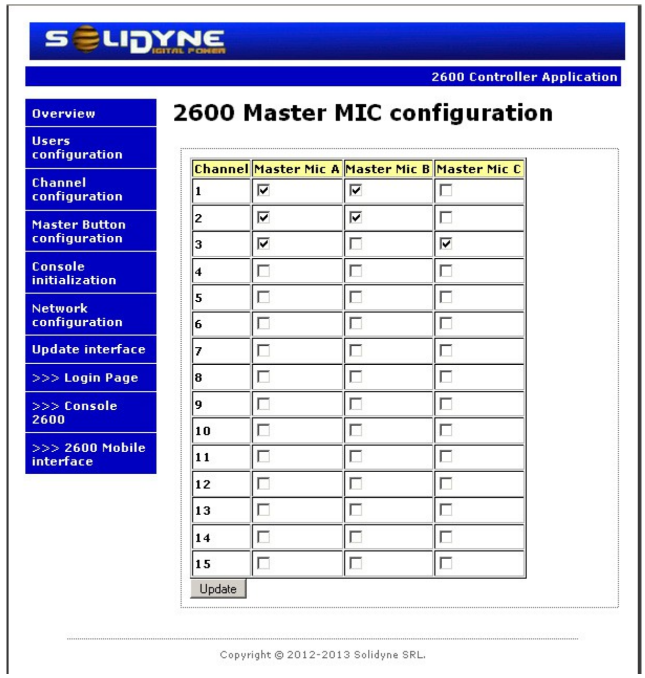
The "User Settings" window also allows to modify the password for the Administrator user. The user name "Admin" can not be changed.

3.8.4.2 Channels settings



This option allows to define the name for each channel, the kind of module and the cross-fade times. To confirm the changes click "Update".

3.8.4.3 Groups of microphones (Master Mic)



'Virtual 2600' has 3 buttons 'Master MIC'. The microphone channels can be assigned to any of the Master MIC buttons. **This setting is common to all users.**

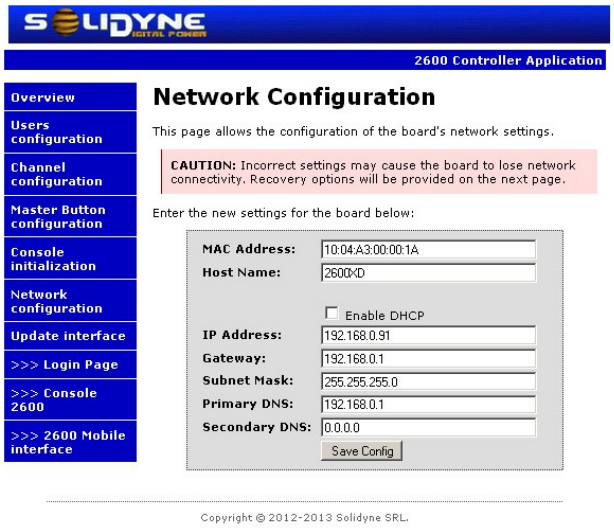
3.8.4.4 Initialize the console

Option	Description	Action
#1	Inicializa los modulos agregados.	Bus Enumeration New modules
#2	Inicializa todos los modulos	Bus Enumeration All
#3	Actualizar info de modulos	Update info
#4	Establecer la fecha y hora de la consola	Set Time

"Listing" and "Update info" are functions that are used for the software to recognize changes in hardware, for example if new modules are added. The procedure is described in the software.

The 'Set Time' option sets the console clock with the current time of the computer. When 'Virtual2600' access to the console for the first time, the clock on the console is set to the time of the computer, even though it had already been set manually from the console.

3.8.4.5 Network configuration



Default the '2600 'works with Dynamic IP (DHCP), so that the router assigns an IP available. Can be assigned a static IP in the "Network configuration" option.

Maintenance — Chapter 4

In order to obtain great results with the 2600 consoles that Solidyne guarantees with its design and manufacture, we recommended following the indicated operative procedures in this manual.

4.1 Fuses

The unit have a **main fuse** (1A) located at rear panel.

The **phones lines** have a circuit to protection against discharges of high voltage. This circuit is mounted in a separated plug-in board, to facilitate its replacement in case of damage.

4.2 On console's temperature

The rear panel of the VU-meters turret is made in aluminum and dissipates the heat produced by the power supply and audio amplifiers.

In models XS and XL the value of overheating with respect to the atmosphere is of around 68º F. That is to say, with a normal atmosphere of 77º F the back panel normally will work at 113º F.

In models XL with 16 channels working with 110 V AC (the extreme condition) the temperature can reach 131º F (with 77º atmosphere) in the back panel. This it is a safe value and the console is designed to work at these temperatures, so you don't worry about it.

4.3 Cleaning

Maintain the room clean and free of dust. The surface of the console can be cleaning using a very smooth detergent (like are used for painted walls) and a sponge or fine cloth hardly humid. NEVER USE alcohol, benzene or petroleum derivatives.

Take this rule: NO SMOKING at the control room. The cigarette ashes are LETHAL for the faders and affect, in addition, to other equipment of the radio (CD players, minidisks, etc.). By the same reason, don't drink or eat while you are working on the equipment.

If after using the console for a long time, you note that the faders become difficult to handle (hard to move) or fails, probably they are dirty. In order to clean it consults with the technical department of your radio. They will have to use special products. If after cleaning the faders the faults remain, consults with your Solidyne dealer to acquire the corresponding spare parts.

Polycarbonate plastic protection: Solidyne consoles use all the front panel modules covered by a hard polycarbonate sheet painted and labeled from the reverse side. This unique protection technology allows for a full protection from hands abrasion. The 2600 console will remains unaltered after a full life of operation.

4.4 Preventive maintenance

The preventive maintenance is carried out during the normal operation of the console to avoid flaws. The 2600 console is manufactured using hi-tech integrated circuits and heavy-duty electronic components, that guarantee an excellent reliability and allows eliminating the routines of preventive maintenance.

The linear faders manage only DC current. Under normal conditions, the useful life of these faders exceeds 200.000 full cycle operations. Use carefully: don't' kick the faders and don't' push them with excessive force.

Special models are provided with the new ESF electrostatic fader technology, without moving contacts, that lasts 25 years.

4.5 Parts Replacement

All modules are assembled using connectors. The inputs modules can be disconnected and reconnected while the console stays on the air (Hot Swap technology). The main faders have connectors too. They are mounted to the chassis with two screws, so its replacement is very easy.

4.6 Service Manual

You can obtain a service manual of this equipment, free costs. For it the Director of the radio must send by FAX to Solidyne a signed Agreement of Confidentiality (the text is available in our Web). You will receive a link to download the service manual, which include schematics; components layouts and technical information. The document installs in a PC that will be used in the Technical Office of the radio. Downloaded manual only can be viewed in that PC.

For more details please consult www.solidynePRO.com. Go to English > Manuals > Service Manuals.

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Specs & Measurements — Chapter 5

5.1 General Measurements

The radio stations that have their own Engineering Department sometimes needs to carry out measurements when receiving the console. Also some engineers estimate convenient to carry out every five years a general inspection of the console to verify if the specifications continues being perfect.

The following methods and comments refers to Technical Specifications that figure at the end of this chapter.

Before starts any measurement, make sure that all modules present the following conditions: PAN POT at the center position. PGM, AUD, and SEND buttons released; AIR and CUE must be off. Be sure the console is **properly grounded** and no RF is present at the measurement Laboratory.

5.1.1 Microphone

Connect an audio generator to a microphone input. Connect an audio level meter and an oscilloscope to the PGM left output (then repeat with the right). Connect a 600 ohms charge to the output.

Set the audio generator to 1 kHz -80 dBm. Select MIC and PGM on the channel which the generator is connected. Enable the channel pressing the AIR button. Move the main fader from this channel to the maximum. Increase the gain of the module until obtain +4dBm at the output.

Move GAIN to the minimum. Change the generator to -45 dBm. Move the fader until you verify that can obtain +4 dBm without clipping at the output.

Change to the right input of this channel (or to another module) and repeat the procedure.

5.1.2 Line

Connect the generator to left input of a line channel, with a level of -20 dBm at 1 Khz. Verify that the oscilloscope and audio level meter are connected to the left PGM output. Select LIN and PGM in the channel under test. Enable the module by pressing the Air button; move to the maximum the GAIN preset and move the main fader until you verify that can obtain +4 dBm at the output.

Change the gain control to the minimum; and increase the input level up to +18 dBm; move the fader until obtain +4 dBm at the output without visible clipping.

Increase the gain with the main fader until obtaining +15 dBm at the output. Use this value like reference. Connect the balanced input in common mode joining both signal terminals. Then, verify that the output level decrease at least 40 dB. Change the test frequency to verify the common mode rejection specification.

Repeat for the right input of this channel or for another module.

5.1.3 Aux Input

Connect the generator to the AUX left input. Select AUX and repeat the procedure explained for the line channels, with levels of -25 dBm and +4 dBm. Take in mind that the common mode rejection isn't applicable for this input.

5.1.4 Balanced Outputs

All the measurements must be carried out in the same way that the unbalanced outputs, but disconnecting of GND the instrumental used and connect it between the two balanced terminals.

Another possibility is to measure all in unbalanced mode (only one pin), **adding 6dB to the results.**

5.1.5 Unbalanced Outputs

Connect the audio generator to line left input; adjusted at +4dBm/1KHz. Connect an audio level meter and the oscilloscope with a charge of 600 ohms, to the SEND left output (pin 9 of the master connector).

In the channel under test, select LIN and SEND. The other buttons must be out.

Enable the channel by pressing the AIR button. Open the main fader up to -10 dB. Change the generator level until reach the limit of clipping. Verify that the oscilloscope shows a greater or equal level +18 dBm.

Repeat for the right channel (terminal 10 of the Master connector). In the previous condition, repeat the output measurement for mono PGM output.

5.1.6 Gain

Connect the microphone input to an audio generator. Connect the audio level meter and the oscilloscope to the output. Charge this output with 600 ohms.

Adjust the generator to 1 kHz / -80 dBm. Select MIC and PGM in the channel. Enable the module by pressing the AIR button. Take the main fader and the gain control to the maximum position. The difference between the obtained output level and -80dB is the gain of the console.

5.1.7 Frequency Response

Connect the generator to a microphone input. Connect an audio level meter and an oscilloscope to the output. Load it with 600 ohms. Change the audio generator to 1 kHz / -50 dBm. Select MIC and PGM in the channel. Enable the module pressing the AIR button. Move the main fader to the value -10 dB. Change the gain of the module or the generator output until obtaining +4dBm at the output. Change the frequency between 20 and 20.000 Hz and verify the frequency response.

5.1.8 Phase

Staying the conditions of the previous item, connect the generator to both microphone inputs (right and left). Connect a digital phase meter to the channels left and right of the program output. Load each channel with 600 ohms. Change the generator's frequency to measure the phase. The variation will be smaller than 2 degree between 50 Hz -15 KHz.

5.1.9 Stereo Tracking

Maintain the conditions of the previous item. Change the generator frequency to 1 kHz. Take the main fader to the maximum. Adjust the Pan-Pot to obtain the same level in both channels (L & R) of PGM output. This level will be in order of +10 dBm. Now, move the fader between 0 and -30 dB and measure the difference among the levels of both program outputs. This difference will be below +/- 0.2 dB

5.1.10 Harmonic Distortion (THD)

Connect the audio generator at 1 KHz / + 4 dBm, to a left line input. Connect a Harmonic Distortion meter and the oscilloscope to the left program output. Load this output with 600 Ohms. Then, select LIN and PGM in this channel. The other switches should be out.

Enable the module by pressing the AIR button. Take the fader to the maximum level (0 dB). Then, change the preset line level until obtaining +4 dBm on the left output. Measure now the total harmonic distortion. Change the frequency between 30 and 15,000 Hz and to check if the distortion is below the specification. Repeat for right channel.

Reduce now the level from the generator to -50 dBm and connect it to the microphone input. Take the fader to 0 dB position. Select MIC; change the level from the MIC preset gain, until obtaining + 4 dBm at the output and to proceed like in the previous item.

> It is necessary to keep in mind so that this measurement has validity, the following conditions must be verified:
>
> **1.** The measurement chain must have a distortion smaller than 0,002%.
>
> **2.** The distortion components, just as they are seen at the oscilloscope screen, connected to the output of the THD meter, must be clearly distinguished from the residual noise and buzz.

5.1.11 Equivalent Input Noise

Connect the audio generator to a microphone input. Load it with 600 ohms. Connect to PGM output an audio level meter with A-weighted filter. Change the generator output to 1KHz / -45 dBm. Select MIC and PGM in the channel under test. Enable the module by pressing AIR. Set the main fader to -10 dB. Change the gain control until obtaining +4 dBm at the console output (this is the reference level: REF).

Now, replace the audio generator by a resistor of 150 ohms placed inside the D-25 MIC input connector. Measure the residual noise in the audio level meter with “A-weighting” filter. We will denominate it Vn(dBm). Verify in the oscilloscope that there is not any buzz; only random noise signal. In order to eliminate buzz, reconnect the grounds of the measurement instruments so that the buzz disappears. The level of equivalent input noise will be:

EIN = Vgen + Vn-REF; that is to say:
EIN (dBm) = 49 + Vn (dBm)

Waited value: 133 dBm

✍ If the “A- weighted” filter is not available, a simple RC filter will be used, that attenuates 3 dB in 15 KHz. The measured noise will be between 5 and 7 dB above to the real one.

5.1.12 Signal/Noise ratio

In the same outline of the previous point, connect the audio generator at 1 kHz / + 4 dBm to the line input. Connect to the output an audio level meter with “A” weighting filter and an oscilloscope. Load the output with 600 ohms.

Move the fader to -10dB position. Adjust the line level preset until get +4dBm output and use this value as reference for the measurement of noise. Replace the generator by a resistor of 600 ohms and measure the new output level. This value plus + 4dBm is the S/N ratio.

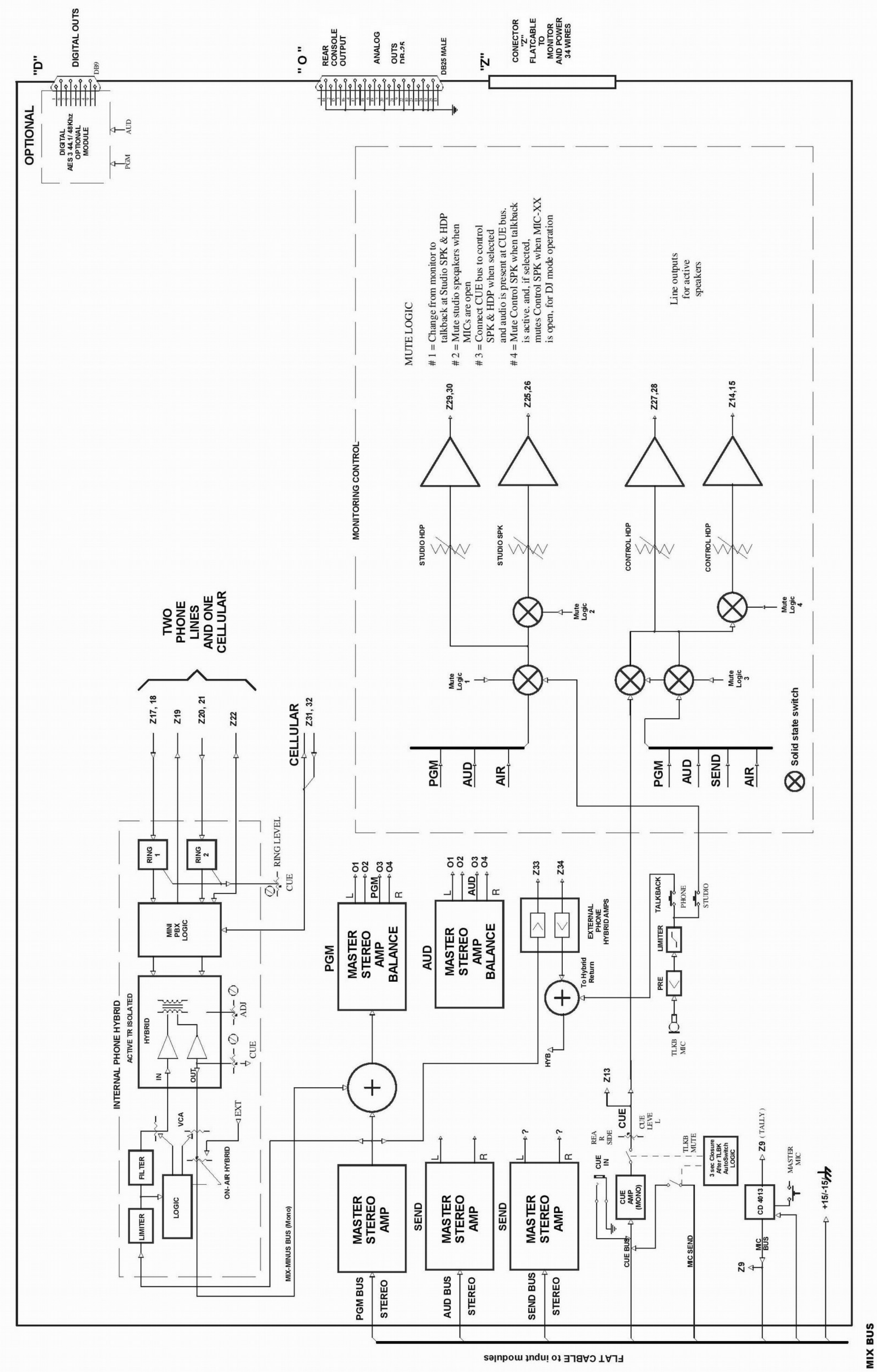
✍ This measurement result is the noise level in dBA.

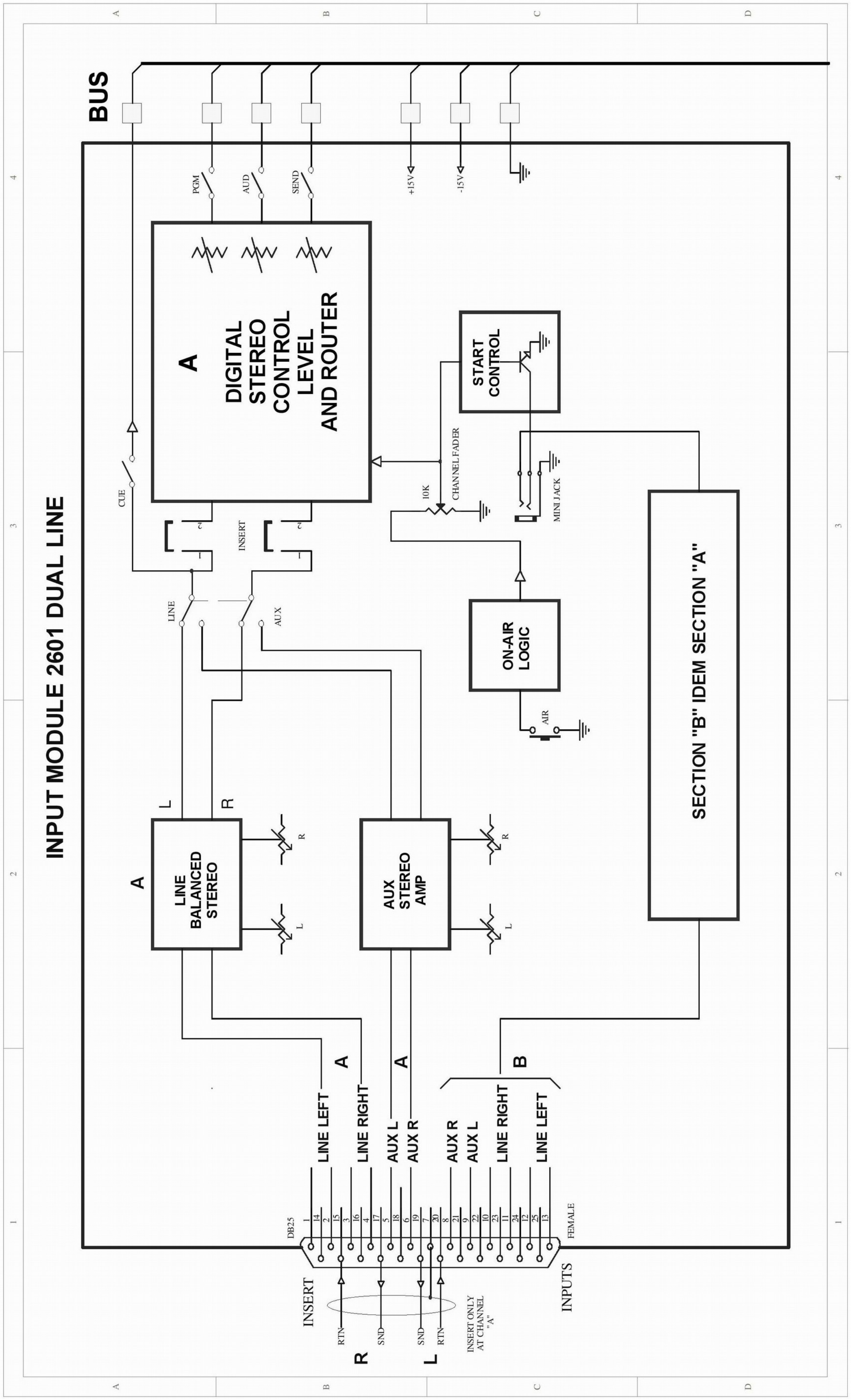
5.1.13 Crosstalk

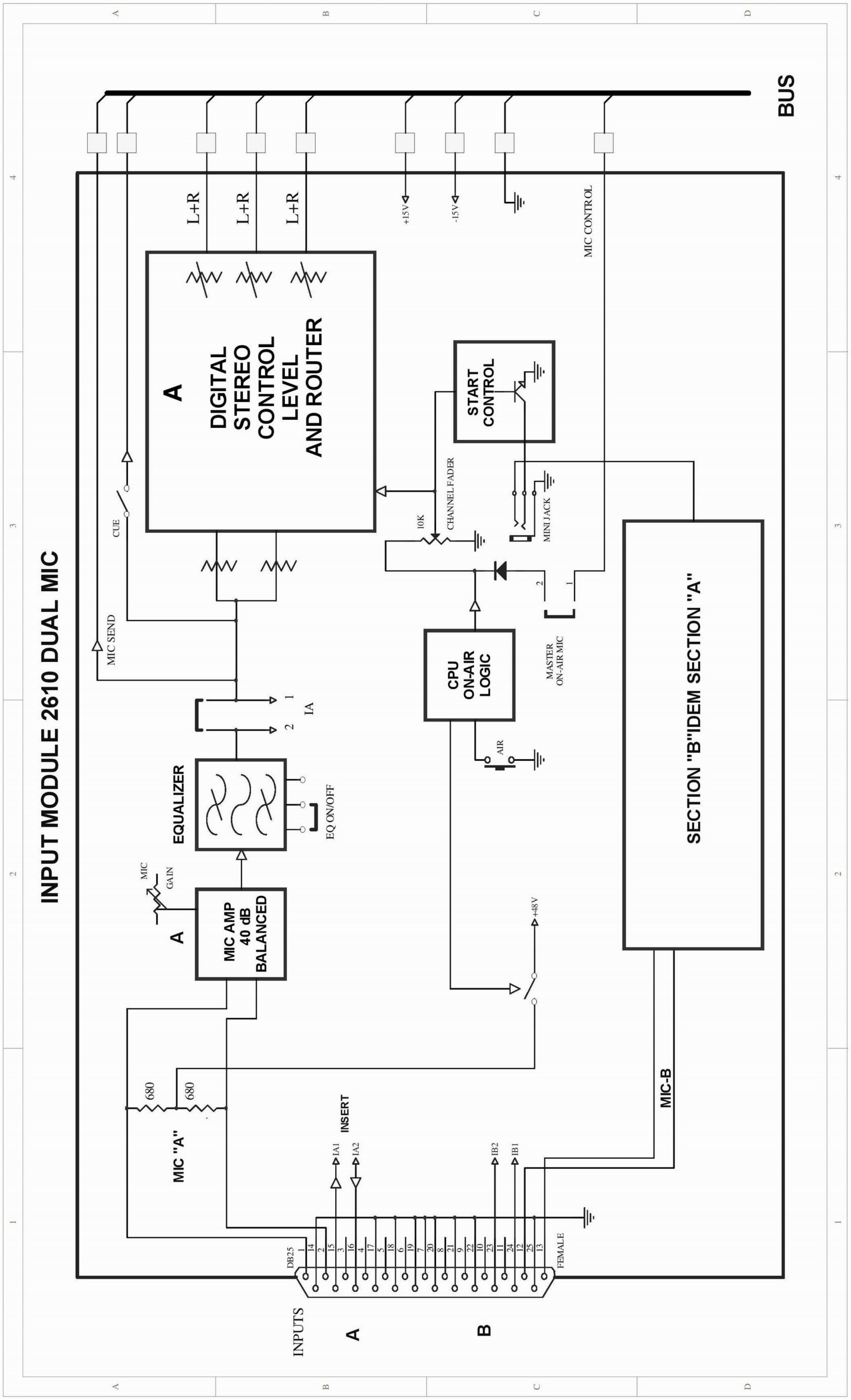
Connect the audio generator (+4 dBm; 1kHz), to an left channel of a line input. Connect an audio level meter with “A” weighting filter, an oscilloscope and a load of 600 ohms to the left **AUD** output. Connect another load of 600 ohms to the left **PGM** output. Select all buses (LIN, PGM and AUD) in the module. All buses in others modules must be out. Enable the module pressing AIR. Move the fader to the maximum. Change the level of the generator until obtaining +15dBm in the AUD output. Change the position of the audio level meter from AUD to PGM output. Verify that the audio level is +15 dBm +/- 0,5 dB. **Release the PGM button** and measure the residual level of the signal. This level, referred to +15 dBm, is the crosstalk between left audition and left program. Repeat for all the combinations of left and right program with left and right audition. In the same way, the crosstalk on PGM bus can be measured. Check it with the console specifications.

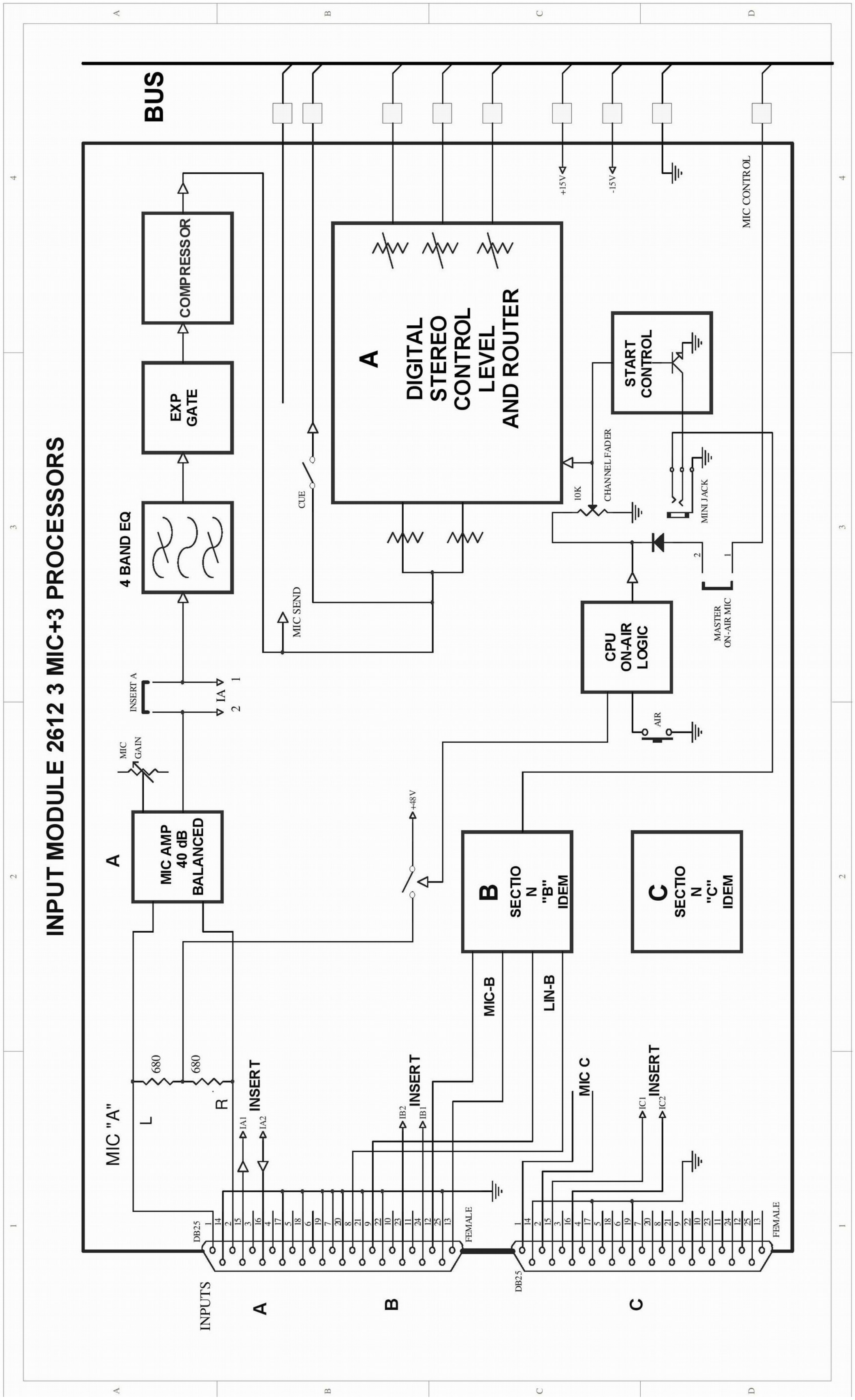
5.2 Bock diagrams

MASTER MODULE 2607 (Block diagram)




INPUT MODULE 2601 DUAL LINE
BUS
PGM
AUD
SEND
+15V
-15V
A
DIGITAL
STEREO
CONTROL
LEVEL
AND ROUTER
START
CONTROL
CUE
INSERT
CHANNEL FADER
10K
MINI JACK
LINE
AUX
ON-AIR
LOGIC
AIR
SECTION "B" IDEM SECTION "A"
A
LINE
BALANCED
STEREO
AUX
STEREO
AMP
L
R
A
A
B
LINE LEFT
LINE RIGHT
AUX L
AUX R
AUX R
AUX L
LINE RIGHT
LINE LEFT
DB25
FEMALE
INSERT
INPUTS
RTN
SND
SND
RTN
R
L
INSERT ONLY
AT CHANNEL
"A"


INPUT MODULE 2610 DUAL MIC
BUS
L+R
L+R
L+R
+15V
-15V
MIC CONTROL
A
DIGITAL
STEREO
CONTROL
LEVEL
AND ROUTER
START
CONTROL
CHANNEL FADER
10K
MINI JACK
CUE
MIC SEND
IA
EQUALIZER
EQ ON/OFF
MIC
GAIN
A
MIC AMP
40 dB
BALANCED
CPU
ON-AIR
LOGIC
MASTER
ON-AIR MIC
AIR
+48V
SECTION "B"IDEM SECTION "A"
MIC-B
680
680
MIC "A"
INSERT
IA1
IA2
IB2
IB1
DB25
FEMALE
INPUTS
A
B


INPUT MODULE 2612 3 MIC+3 PROCESSORS
BUS
MIC AMP
40 dB
BALANCED
MIC GAIN
INSERT A
IA
4 BAND EQ
EXP GATE
COMPRESSOR
MIC SEND
CUE
DIGITAL
STEREO
CONTROL
LEVEL
AND ROUTER
+15V
-15V
MIC CONTROL
START CONTROL
10K
CHANNEL FADER
MINI JACK
MASTER ON-AIR MIC
CPU
ON-AIR
LOGIC
AIR
+48V
B
SECTIO
N
"B"
IDEM
C
SECTIO
N
"C"
IDEM
MIC-B
LIN-B
MIC C
MIC "A"
L
R
680
INSERT
IA1
IA2
IB2
IB1
IC1
IC2
DB25
FEMALE
INPUTS
A
B
C

5.3 Technical specs

Audio Inputs

A & B – 2 inputs per channel. 2601 balanced stereo line and unbalanced AUX. 2610/2612 balanced MIC
2620 Digital USB and balanced AUX.

In option USB 2602, 2 inputs and 2 outputs

Outputs

3 bus outputs; PGM & AUD balanced + 4 dBm;
AUX Unbalanced + 4 dBm
Balanced Outs; Max Level +26dBu (10K), +20dBm (600ohms)
Unbalanced Outs; Max Level = +20dBm/600 ohms
Optional Digital AES/EBU output for PGM & AUD.

Inputs Level/Impedance

Balanced MIC= -25 dBu/-80 dBu; 150/250 Ohms
Balanced LINE = -20 dBu/+22 dBu; 600~ 10Kohms
Unbalanced AUX = -20 dBu/+15dBu; 600~ 10Kohms

Digital Levels

AES-3/SPDIF outputs TCP/IP & USB inputs & outputs; 0VU RMS is set at -15dB Ref Full Scale level. 0dB Peak is set to 100% Full Digtal Scale

Phantom 48v

Standar phantom 48V power supply for up to 10 microphones.

Monitor & Hybrid Outputs

4 stereo monitor outputs for active speakers and headphones, including internal hedphones distribution Amp (up to 8 headphpnes) in announcer studio.
1 External Hybrid input + 4 dBu/10K
1 External Hybrid send output (MIX-Minus) +4 dBu / 10K
Each 2602 channel has one Mix-Minus out at 0 dBu output

CUE Monitor

3W CUE monitor with internal speaker.

Phone Hybrid

3 Hybrid inputs, 2 for phone lines and one for cellular phone at 4 wires. It includes a mini-PBX with Ringer, Line attention with free hands operation, and Line Transfer.

Frequency Response: 250 - 4.000 Hz
Noise: > 60 dBA S/N
Rejection: > 30 dB rejection
Rejection adjust:: Quick Test Mode to adjust the hybrid balance without disturbing On-air operation.
Preference attenuation: 10 dB local speaker interrupt priority.
Automatic On-Air logic: Audio & Logic are managed from a single 100mm slide fader that performs all the operations in error-free mode.
Lighting discharge safety: Telephone Hybrid inputs are transformer floating to meet the Public Telephone service isolation standards. They are protected with SIOV Varistors against lightning discharges. And are factory tested to 2.000 volts capacitor discharge according to Siemens/Ramatel 44.04 standard. Protector is mounted in plug-in board. We recommend in some areas using external gas discharge protection

Voice Quality Restoration

The optional 2630 module allows using VQR technology for all the telephone hybrid signals, included external hybrids. It means to restore the cuasi-original voice quality that is impaired during a telephone or cellular communication.

ON-AIR LIGHT

ON-AIR signal output (turns on when MIC is open) 12 VDC @ 0,3 amp

Start external devices

One Start Devices output for each input module, to start external devices when open each channel fader. Out at minijack 1/8" // Ring = A channel (left side) Tip= B channel (Rigth side) Open collector OFF= Open. Manages 0-24 VDC @ 100 mA. This allows to control ***Virtual Rack*** processors and start audio devices.

Headroom

22 dB @ LIN a PGM Ref + 4dBu/10 k

Frequencies response

20-20.000 Hz +/- 0.25 dB (LIN or MIC to PGM)

Noise

MIC, EIN=-132 dBu/200 ohms
LINE, S/N > 95 dBA

Dynamic range

From LINE to PGM > **110dBA** (better than “CD quality”)

Crosstalk

PGM-AUD-SEND > 90 dBA @30-10.000 Hz
L-R & R-L > 80 dBA @ 1khz

Distortion

From LINE to PGM out < 0.01 % THD @30-15.000 Hz.
From MIC to PGM out < 0.01 % THD @30-15.000 Hz

Phase

From Line to PGM, < 2º L&R @50-10.000 Hz

Stereo tracking

Below 0.2 dB error L/R in fader range 0 to 40 dB

Talkback microphone

Included Talkback MIC, with Audio Limiter. Noise Cancelled
PZM type / Outputs to Phone Line Hybrid and Studio Speaker
with AutoSwitch (AutoSwitch allows one-touch Studio communication)

VU-meters

2600XL: 4 digitally controlled VUmeters 24 steps of 1 dB,
DUAL type (RMS & Peak) for PGM & AUD. One Timer/Clock and one Stereo Phase Vector display

Timer /Clock Option

Timer counts minutes: seconds during on-air MIC in order to avoid excessive speaker on-air time When the Timer is off, a standard clock is activated

Power source

40W switch 90-240V , 50/60 hz, CE, UL, TUV Certified

Dimensions & Weight

XL model: 840 wide x 585 depth x 135 height - Weight 15 Kg
BOX : 90x65 x 25 cm, 2 Kg

2600XD: 845mm wide, 635mm depth, 125 mm height - Weight 15 Kg
BOX: 90x70 x 20 cm, 2 Kg